iPECS

Feature Operation
&
Description Manual

LG-ERICSSON

Regulatory Information

Before connecting the iPECS to the telephone network, you may be required to notify your local serving telephone company of your intention to use "customer provided equipment." You may further be required to provide any or all of the following information:

PSTN line Telephone numbers to be connected to the system	
Model name:	LIK-MICRO/100/300/600/1200 IP KTS
Local regulatory agency registration number:	locally provided
Ringer equivalence:	0.7B
Registered jack:	RJ-11 with Desk Holder/Wall Mount RJ-21X with Main Cabinet

The required regulatory agency registration number is available from your local LG-Ericsson representative.

This equipment complies with the following regulatory standards, FCC Part 15 and 68, IC (Industry Canada) CS03, TBR21, TBR03, and TBR04. Also, this equipment complies with the safety requirements of UL60950, CSA60950, EN60950, EN55022 and EN55024.

The iPECS has been designed to comply with the Hearing Aid Compatibility requirements as defined in Section 68.316 of Part 68 FCC Rules.

If the telephone company determines that customer provided equipment is faulty and may possibly cause harm or interruption in service to the telephone network, it should be disconnected until repair can be affected. If this is not done, the telephone company may temporarily disconnect your service.

The local telephone company may make changes in its communications facilities or procedures. If these changes could reasonably be expected to affect the use of the iPECS or compatibility with the network, the telephone company is required to give advanced written notice to the user, allowing the user to take appropriate steps to maintain telephone service.

The iPECS complies with rules regarding radiation and radio frequency emission as defined by local regulatory agencies. In accordance with these agencies, you may be required to provide information such as the following to the end user:

WARNING

This equipment generates and uses R.F. energy, and if not installed and used in accordance with the Instruction Manual, it may cause interference to radio communications. It has been tested and found to comply with the appropriate limits for a telecommunication device. The limits are designed to provide reasonable protection against such interference, when operated in a commercial environment.
Operation of this equipment in a residential area could cause interference, in which case the user, at his own expense, will be required to take whatever measures may be required to correct the interference.

CAUTION

This system employs a Lithium battery as back-up power for the real-time clock and memory. The battery is not replaceable in the field. Dispose of used batteries accordance with the manufacturer's instructions.

Revision History

ISSUE	DATE	DESCRIPTION OF CHANGES
1.0	20-Mar-02	Initial Release
1.1	21-Aug-02	General Update
1.2	8-Nov-02	Update for Software version 1.1Bd
2.0	22-Sep-03	Update for release 2 software
2.1	16-Feb-03	Update for European release 2 software
2.2	29-Jul-04	Update for Software version 2.0Ai
2.3	12-Dec-04	Draft for version 3.
3.0	22-Feb-05	Update for iPECS S/W Phase 3 (3.0As base)
4.d2	24-Aug-06	Updates for iPECS release 4 software and LG-Nortel logo General edits through-out. Changed formatting (header, footer, etc.) for unified iPECS Manual appearance. CLI based Call Routing for DID, added reference to DID Lines in ICLID routing. Flexible configuration for up to 600 ports, covered in section 1.2. Hunt Group Mailbox, referenced mailbox assignments in Station Hunt Groups. Multiparty voice conferencing (Up to 24 Parties in total), added to Multi-Party Conference. Conference Room Added Conference Room. Simplified Message retrieval added to Message Wait/Call Back and restructured. Muted Ring w/Pre-Selected Msg added & restructured Pre & Custom Display Messages. T.38 Fax Relay, added IP FAX, T.38. Day/Night Mode applied between groups, added reference and rewrote ICM Tenancy. ACD sections previously section 2.5 to 2.20, restructured ACD sections now section 2.5.1~2.5.12. Multi-level Admin access, restructured admin with Multi-level section. Programmable Admin TCP port, restructured Web Admin & added programmable tcp port. Web User manual, multi-lingual, added Web User Manual section with Multi-lingual support. Network Security & Priority added new section with VLAN, Diffserv, IPSec and SRTP. Transparent Networking added Centralized Networking with Fail-over & Remote Gateways. Hunt Group Station Forward, added condition for Hunt group recognition of forward state. System Processor Redundancy, added section for standby MFIM600. Call Recording, added unconditional Call recording feature. IP Trunking, added SIP and H.323 with GK routed call VSF section revised to VMIM/VSF, added VMIM description. VMIM e-mail notification added VMIM Back-up station added VMIM multi-language support MOH, added VMIM/VSF MOH Authorization codes, added new operation for system codes, "* + table index + code". Mobile Ext. enhanced, mobile receives hunt calls and support for Attd. Recall.
4. d3	30-Sep-07	Updates for iPECS release 4 software (General edits for errata). Emergency Call Attendant Alert Automatic Daylight Savings Time (DST) Adjustment Direct Inward Dialing (DID) Automatic System Time Synchronization Distributed Systems Network, paging E-911 support for Emergency dialing VSF/VMIM Outbound Notification to Pager or External Phone TLS for Web Admin and Webphone
5.0	26-July-08	General updates & errata Change "MFIM" references to "iPECS, include iPECS 50 and updated capacity table Reference to VSF/VMIM announcement changed, expansion from 20 to 70 codes, VSF MOH code 21 or 71) Multi-language code 22 or 72 throughout.

		Updates for iPECS release 5 software One-time DND forward For TNET added paging reference and RTP description. Device Zone Management Call Coverage enhancement SIP Extension support Tenancy Group ring mode for Flexible DID/ICLID Station ICR Conference Group w/recording) iP Bridge Cabinet Alarm NMS System DECT DISA Call Forward removed Added LIP-8000 series soft key operation for auto dial features (LNR, SAVE, etc. Added Attendant active MFIM display Updated Table 1.2-1 and updated VSF capacity Call Forward using external number added Added Option for voice mail attachment to e-mail Change BGM operation SMDR ICM added
5.0d1	07-Jan-09	Add in capacity table and Changed for iPECS-1200
5.0d2	17-Feb-09	Add group name for Terminal/Circular/VM/UCS.
5.0d3	13-Apr-09	Add condition 3 in 2.72.3.6 (Voice Mail Back-up Station).
5.0d4	12-May-09	Updatet for outgoing mailbox dsstination
5.0d5	02-Feb-10	Update description of DID call wait feature.
5.5	09-JUL-10	Updates for iPECS release 5.5B software Add IP Watch timer for cpu redundancy and IPCR
5.5d1	30-DEC-10	Updates for iPECS release 5.5C software
		AU Mods: Manual Application, Enter Conf Room *59, Conf Grp *68, Group Pickup **, Paging codes *101-*110,#3,##,#41,#5,#00, Call Coverage *67, Directed Call Pick-Up *7, Attendant 9, CO Access 0, Unsupervised Conf Timer Extend *##, AOC N/A, Emergency Call E-911 N/A
5.5d2	30-JUL-11	Updates for iPECS release 5.5D software
	09-MAY-12	Add missing program for Auto Call Recording & Two-way record.

Table of Contents

1. INTRODUCTION

1.1 MANUAL APPLICATION

This document provides detailed information covering description and operation of the numerous features available in the iPECS Series Release 5.5 system software. As with any instruction the suitability and correct application needs to be assessed on a site by site basis. Thus instructions below may need to be adapted accordingly. Due to a number of reasons, eg parts, quality and type of CO lines, etc, features below may be unsuitable. In the event that a customer finds the application of a solution unsuitable please take the necessary steps to rectify. To rectify an application that is not suitable one may need to reverse the activation steps and may include turning the application off. Many features have constraints and limitations documented in these technical manuals. In many cases some general limitations are not documented. Non publication of a limitation does not constitute responsibility on Aria Technologies part to meet customers expectations unless deemed as a core advertised function failing in a standard setting e.g. Incoming ring, outgoing call access. The document is written assuming the system employs the default-numbering plan for North America (adjusted to AU for most values).

1.1.1 Organization

Features are arranged alphabetically in six different major groupings as follows:

- Section 2 System Features
- Section 3 Intercom Features
- Section 4 CO/IP
- Section 5 iPECS Phone
- Section 6 Attendant Position
- Section 7 SLT

1.1.2 Feature Information

Each section is an alphabetical listing of features with the description and operation of each. The structure is divided into 6 parts as below:

- **Description:** explains the nature of the feature.
- **Operation:** gives detailed step-by-step operation of the feature for iEPCS IP Phones and SLTs.
- **Conditions:** explains known feature interactions and constraints related to the feature.
- **Programming:** lists database entries that may be required for proper feature operation.
- **Reference:** lists related topical information to aid in understanding the feature.
- **Hardware:** lists hardware required for proper feature operation.

1.2 SYSTEM CAPACITIES

The iPECS Series is available in several configurations as listed in Table 1.2-1. Total port capacities range from the 31-channel iPECS-Micro to iPECS-1200 at 1200 ports.

Table 1.2-1 System Capacity Chart

Item	Capacity

	iPECS-Micro	iPECS-50	iPECS-100 (MFIM100)	iPECS- 300 (MFIM300)	iPECS-600 (MFIM600)	iPECS-1200 (MFIM1200)
Main Cabinet	n/a	n/a	10 slots	10 slots	10 slots	10 slot
System Ports	31 ports	50 ports	100 ports	300 ports	600 ports	1200 ports
Stations*1	26 w/2 SLT std.	50 w/2 SLT std.	70	300	600	1200
PSTN circuits*[1]	5	max 42, basic **iPECS 50A** std. 4 PSTN + VoIP **iPECS 50B** std. 4 BRI + VoIP	42	200	400	600
Max. RSGMs*[2]	13	25	35	150	300	600
Attendants	4	4	4	5	5	5
Serial Port (RS-232C)	n/a	1	1	1	1	1
USB Host port	1	1	1	1	1	1
Alarm/Door bell input	n/a	1	2	2	2	2
External Control Relays	n/a	1	2	4	4	4
Music Source Inputs	n/a	1	2	2	2	2
Power Fail Circuit	n/a	1	4	4 + Ext. PFTU (6 optional)	4 + Ext. PFTU (6 optional)	4 + Ext. PFTU (6 optional)
External Page zones	n/a	1	2	2	2	2
Internal Page Zones	10	10	10	35	35	100
System Speed Dial	800 (48 digits)	800 (48 digits)	800 (48 digits)	3000 (48 digits)	6000 (48 digits)	12000 (48 digits)
System Speed Dial Zones (Groups)	10	10	10	10	20	50
Station Speed Dial	20 (48 digits)	20 (48 digits)	20 (48 digits)	100 (48 digits)	100 (48 digits)	100 (48 digits)
Total station speed dial	1000	1000	1000	4000	8000	24000
Last Number Redial	10 (48 digits)	10 (48 digits)	10 (48 digits)	10 (48 digits)	10 (48 digits)	10 (48 digits)
Save Number Redial	1 (48 digits)	1 (48 digits)	1 (48 digits)	1 (48 digits)	1 (48 digits)	1 (48 digits)
DSS Consoles/Station	1	3	3	9	9	9
SMDR buffer	5000	5000	5000	10000	15000	30000
CO Line Groups	20	20	20	72	72	200

Item	Capacity					
	iPECS-Micro	iPECS-50	iPECS-100 (MFIM100)	iPECS- 300 (MFIM300)	iPECS-600 (MFIM600)	iPECS-1200 (MFIM1200)
Station & Hunt Groups	12	40	40	48	48	100
Station & Hunt Group Members	26	50	70	70	70	200
Pickup Groups	20	20	30	100	150	200
Pickup Group Member	26	50	70	300	600	1200
Conf. Grps - System	20	20	20	40	80	160
Station	25	25	35	150	300	600
Executive/Secretary pairs	10	10	10	36	36	100
Authorization Codes						
Station	26	50	70	300	600	1200
System	474	450	430	700	1400	2800
Total	500	500	500	1000	2000	4000
VSF[*3]	280(245) minutes (4 channels)	280(245) minutes (6 channels)	210(175) minutes (6 channels)	210(175) minutes (6 channels)	n/a	n/a
VMIM	n/a	9 hrs	9 hrs	9 hrs	9 hrs, max x 6	9 hrs, max x 30
MCIM	1	2	2	4	8	8
WTIM	16	16	16	32	32	32
VoIP channels[*4]	5 std.	Std. 4-8	6 std.	6 std	n/a	n/a
Redundancy	No	No	Yes	Yes	Yes	Yes
SIP channels (Stations + CO lines)	Same as system port	Same as system port	100	200	200/100 Note5	600

Note 1 *The station and CO Line maximums are not simultaneously; total ports cannot exceed the specified System Port capacity.*

Note 2 *For maximum RSGM connection ports, calculation formula is ports = available system station ports)/2, there must be sufficient VoIP channels to support packet relay for RSGM rtp packets.*

Note 3 *Approximately 35 minutes (16 Mbytes) of the VSF memory are used to provide basic system prompts, the remaining memory can be used for announcements and voice message storage. Values in parenthesis are the announcement and storage time available.*

Note 4 *Using G.711 codecs, 8 VoIP channels are available. Due to additional processing needs, complex codecs reduce the available channels; four (4) channels are available using G.723 or G.729.*

Note 5 *Issue 0, 1, 2: Support up to 100 channels (SIP extension + CO trunk) simultaneously*

Issue 3 : Support up to 200 channels (SIP extension + CO trunk) simultaneously

2. SYSTEM

2.1 ACCOUNT CODE

Description

Station users may allow tracking of specific calls by entering a non-verified variable length (up to 12 digits) identifier for a call. The identifier or "Account Code" is output as part of the Station Message Detail Record (SMDR) for the call.

Operation

iPECS Phone

To assign a Flex button for ***{ACCOUNT CODE}*** *operation:*

{ACCOUNT CODE} button:

[PGM] + **{FLEX}** + **[PGM]** + '84' + **[SAVE]**

{ONE-TOUCH ACCOUNT CODE} button:

[PGM] + **{FLEX}** + **[PGM]** + '84' + Account code (up to 12 digits) + **[SAVE]**

To enter an Account Code using an ***{ACCOUNT CODE}*** *button prior to placing a call:*

1. Lift the handset.
2. Press the **{ACCOUNT CODE}** button.
3. Dial the Account Code (1 to 12 digits).
4. Press '*', Intercom dial tone is heard.
5. Place the CO/IP call as normal.

Or,

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **{ONE-TOUCH ACCOUNT CODE}** button.
3. Place the CO/IP call as normal.

To enter an Account Code using an ***{ACCOUNT CODE}*** *button during a call:*

4. Press the **{ACCOUNT CODE}** button.
5. Dial the Account Code (1 to 12 digits).
6. Press '*'.

Or,

7. Press the **{ONE-TOUCH ACCOUNT CODE}** button

SLT

To enter an Account Code prior to placing a call:

8. Lift the handset.
9. Dial Flex Numbering Plan code '550'.
10. Dial the Account Code (1 to 12 digits).
11. Press '*'.
12. Place the CO/IP call as normal.

13.

To enter an Account Code during a call:

14. Momentarily press the Hook-switch.
15. Dial Flex Numbering Plan code '550'.
16. Dial the Account Code (1 to 12 digits).
17. Press '*'.

Conditions

18. When entering an Account Code during a call, DTMF digits are not heard by the connected party.
19. A maximum of 1000 **{ONE-TOUCH ACCOUNT CODE}** buttons may be assigned in the system.
20. If an Authorization Code is entered as the Account Code, the SMDR record will show the station number or the bin number for a System Authorization Code rather than the user entered Authorization Code.

Programming

NUMBERING 1. Flexible Numbering Plan (PGM 106-Btn 8)

Related Features

Authorization Codes (Password)
SMDR (Station Message Detail Recording)
Station Flexible Buttons

Hardware

2.2 ALARM SIGNAL/DOOR BELL

Description

The system can be configured to recognize the status of an external contact (normally open or closed). The system will signal assigned iPECS Phones when the contact activates. This capability is commonly employed to provide remote Alarm or Door Bell signals to the user.

Assigned stations receive the Alarm Signal, either a single tone burst repeated at 1-minute intervals or a continuous tone. The Alarm Signal may be terminated at the user's phone by dialing the Alarm Stop code or, if assigned, pressing the **{ALARM STOP}** button. To rearm the Alarm function, the alarm condition must be cleared and the Alarm signal terminated.

When used as a Door Bell, assigned iPECS Phones receive a single tone burst each time the external contact is activated and no reset is required.

Operation

System

At detection of contact operation, the Alarm/Door Bell signal is sent to assigned stations.

iPECS Phone

*To assign a Flex button as an **{ALARM STOP}** button to terminate the Alarm Signal:*

[PGM] + **{FLEX}** + '565' + **[SAVE]**

To terminate an Alarm Signal while idle:

1. Dial the Flex Numbering Plan code 565, confirmation tone is received and the Alarm Signal is terminated. If the alarm condition is cleared, the system will automatically rearm the alarm monitoring.

Or,

1. Press the **{ALARM STOP}** button.

Conditions

1. The Alarm contacts must be "dry", no voltage or current source connected.
2. Only an iPECS Phone may be assigned to receive Alarm/Door Bell signals.
3. An iPECS Phone with LCD assigned to receive Alarm/Door Bell signals will show "ALARM" or "DOOR BELL" as appropriate.

Programming

SYSTEM

1. Alarm Enable (PGM 163-Btn 1)
2. Alarm Contact Type (PGM 163-Btn 2)
3. Alarm/Doorbell Mode (PGM 163-Btn 3)
4. Alarm Signal Mode (PGM 163-Btn 4)

STATION

1. Alarm/Door Bell Attribute (PGM 113-Btn 8)

Related Features

Door Open

Hardware

iPECS Phone

External contact connected to Alarm input of MFIM, refer to iPECS **Hardware Description & Installation Manual**.

2.3 AUTHORIZATION CODES (PASSWORD)

Description

Authorization Codes provide a means to control access to Off Premise Call Forward, Walking COS, or DISA and may be required for outgoing CO/IP Line or LCR access based on configuration of the iPECS database. When users dial an Authorization Code that matches an Authorization Code stored in the database, the system invokes the Station COS or the COS assigned to Authorization code. Each Authorization code has separate Day/Night mode COS assignments.

There are two types of Authorization Codes, Station and System. A Station Authorization Code is specifically related to a given station and intended for a single user. The System Authorization Codes are intended for use by any station in the system.

The Station Authorization Codes includes the associated station number and the assigned code. The structure of the System Authorization code can be set as either ”*”, or “*” the Authorization table index and the code digits. The later allows duplicate codes to be employed using entry of table index to provide a unique identification of the entry.

The Administrator and Attendants are permitted to assign any Authorization code including codes for another station. Normal users may only assign the Station Authorization code for the specific station.

Operation

iPECS Phone

<u>*To assign a Station Authorization Code:*</u>

1. Press the **[PGM]** button.
2. Dial ‘33’, the Authorization Code Program Code.
3. Dial the Station number.
4. Dial the Authorization Code (1 to 12 digits).
5. Press the **[SAVE]** button.

<u>*To enter an Authorization Code when second dial tone is received:*</u>

1. Dial the station number for the Station Authorization code or, for a System Authorization Code, dial ‘*’ or ‘*’ and the Authorization table index.
2. Dial the corresponding Authorization Code.
3. Place call as normal.

SLT

<u>*To assign a Station Authorization code:*</u>

1. Lift the handset.
2. Dial SLT Program Mode Entry code ‘561’.
3. Dial Station User Program code ‘33’.
4. Dial Authorization Code (1 to 12 digits).
5. Momentarily press the Hook-switch, receive confirmation tone.

To enter an Authorization Code when second dial tone is received;

1. Dial the station number for the Station Authorization code or, for a System Authorization Code, dial '*' or '*' and the Authorization table index.
2. Dial the corresponding Authorization Code.
3. Momentarily press the Hook-switch.
4. Place call as normal.

DISA

To enter an Authorization Code when second dial tone is received:

1. Dial the station number for a Station Authorization code or, for a System code, '*' or '*' and the Authorization table index,
2. Dial the corresponding Authorization Code.
3. Place call as normal.

System Attendant

To assign an Authorization Code:

1. Press the **[PGM]** button.
2. Dial Attendant Station Program Code '031'.
3. Dial station number for a Station code or, for system code, '*' or '*' and the Authorization table index.
4. Dial Authorization Code.
5. Press the **[SAVE]** button.

Conditions

1. When a DISA Line is marked for Authorization Code entry, the caller will hear DND Warning tone and must input a valid Authorization Code to continue. In case of an entry error, the user may retry entry of the code. In case of multiple entry errors, the user may retry entry based on the DISA Retry counter. The station, if assigned, is forced to COS 7 after repeated attempts fail.
2. A user must enter a valid code within the number of attempts assigned as the FAC (Forced Account Code) Retry Count. The station, if enabled, is placed to COS 7 after repeated failure attempts.
3. A user may enter an Authorization Code from any station to place a CO/IP call using Walking COS.
4. The default Station Authorization code is the station number and "*".
5. The total number of Authorization codes is provided in Table 1.2-1.
6. CO/IP groups may be configured to require entry of a valid Authorization Code. In this case, a second dial tone is provided when the CO/IP group is accessed. If the code entered is invalid, error tone is returned and the user must enter a valid code within the FAC Retry Count.
7. If the Check Password option is enabled in the LCR database, when dialed digits match the LDT table digits, the system will send second dial tone to request the user input a valid Authorization code.
8. An Authorization code may include any dial pad digit except '*' and '#'.

9. Duplicate or conflicting System Authorization codes are not allowed when using the older "*" and code operation. For example, code '1234' conflicts with code '123' and cannot be recognized as a unique code. Since the index operation employs the table index and the station number forms part of the Station code, conflicts will not occur and duplicate codes are allowed for these types of Authorization code.
10. Use of Authorization codes varies based on the system nation code. In some regions, particularly the US and UK, a System Authorization code may be required for DISA access. Entering a Station code on a DISA line will fail in these areas. In other regions, a System Authorization code can be used with Walking COS.
11. In case of System Authrization code, there is a system base admin option - End code(#) usage in System Auth Code (PGM 160 – Btn 26).
    If this value is set to ON, user must enter End Code(#) after entering authorization code.

Programming

STATION
1. Station Account (PGM 112-Btn 19)

CO/IP
1. CO/IP Group Authorization (PGM 141-Btn 8)
2. DISA Authorization Code (PGM 142-Btn 5)

SYSTEM
1. DISA Retry Counter (PGM 160-Btn 5)
2. FAC Retry Count (PGM 160-Btn 16)
3. Old Auth Code Use (PGM 161-Btn 16)
4. COS 7 When Auth fails (PGM 161-Btn 17)
5. End code(#) usage in System Auth Code (PGM 160 – Btn 26)

TABLES
1. LCR Check Password (PGM 221-Btn 6)
2. Station Authorization Code Table (PGM 227)

Related Features

Account Code
Auto Service Mode Control
Direct Inward System Access (DISA)
CO/IP Line Groups
CO/IP Access
Temporary Station COS/Lock
Walking COS
Call Forward
Station User Programming & Codes

Hardware

2.4 Auto Call Release

Description

CO/IP or intercom calls (except Hands-free Answerback), will be released automatically if the user does not complete dialing or, for intercom calls, the called party does not answer after a pre-determined time.

Operation

System

Auto Call Release of Intercom calls:

If a station places an intercom call and the called station does not answer in the Intercom Call Release Time, the call is terminated and the calling user receives error tone.

Auto Call Release of CO/IP calls:

If a station seizes an idle CO/IP Line and does not dial within the CO/IP Call Automatic Release Time, the call is terminated and the user will receive error tone.

Conditions

1. If the Automatic Call Release Timer is set to '0', Auto Call Release is disabled.
2. When the handset is used to place a call, the user will receive error tone for 30 seconds followed by 30 seconds of Howler tone and the station is placed in a fault mode. If on-hook dialing is used, the station receives error tone for one (1) second and returns to idle automatically.

Programming

STATION	1. Howling Tone to Stn (PGM 111-Btn 5)
SYSTEM	1. CO Call Automatic Release Timer (PGM 180-Btn 12) 2. Intercom Call Automatic Release Timer (PGM 182-Btn 4)

Related Features

Howler Tone

Hardware

2.5 Automatic Call Distribution (ACD)

Description

iPECS ACD incorporates flexible incoming call routing, real-time agent monitoring and supervision, and call record statistics as well as ACD Event messages for management reporting. Calls route to an ACD group directly or otherwise including via call transfer, CCR and ICLID routing.

2.5.1 Agents

2.5.1.1 Agent Auto Connect/Zap Tone

Description

Agents using a headset can have calls from enabled ACD groups connected to them automatically. This feature removes the requirement for the agent to answer ACD calls manually. The Agent receives a short tone (Zap Tone), if assigned, and is then connected to the ACD caller.

Operation

System

When Zap tone is enabled, operation of this feature is automatic.

Conditions

1. The station must be in the headset mode for this feature to operate.

Programming

Station Groups

1. ACD Group (PGM 191)
2. Zap Tone (PGM 191-Btn 24-1)

Related Features

Station Groups

Hardware

iPECS Phone

2.5.1.2 Agent Id Login/Logout

Description

Stations or 4-digit Agent Ids are assigned as members of an ACD group. A station or Agent ID can be assigned to a maximum of two (2) ACD groups. Each Agent has a unique 4-digit Agent Id used to login and logout of active ACD group participation. Agents can login from any station in the system but only at one station. When an Agent logs in, any active login for the Agent is terminated and the new login is honored. An Agent may login to two (2) groups at one time, logging into and out of the primary and secondary group separately.

Operation

Agent iPECS Phone

*To assign **{LOGIN}** or **{logout}** buttons:*

[PGM] + **{FLEX}** + Login or Logout code + ACD Group Number (optional) + **[SAVE]**

To Login to a primary group using the Agent Id;

1. Lift handset or press **[SPEAKER]** button,
2. Dial '581' the ACD Agent Primary Login code or press the Login Flex button,
3. Dial the ACD group number,
4. Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the primary ACD Group;

1. Lift handset or press **[SPEAKER]** button,
2. Dial '582' the ACD Agent Primary Logout code or press the Logout Flex button.
3. Dial the ACD group number
4. Dial the Agent ID, the agent is logged out of the ACD Group.

To Login to a secondary group using the Agent Id;

1. Lift handset or press **[SPEAKER]** button,
2. Dial '583' the ACD Agent Secondary Login code or press the Login Flex button,
3. Dial the ACD group number,
4. Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the secondary ACD Group;

1. Lift handset or press **[SPEAKER]** button,
2. Dial '584' the ACD Agent Secondary Logout code or press the Logout Flex button.
3. Dial the ACD group number ,
4. Dial the Agent ID, the agent is logged out of the ACD Group.

SLT

To Login to a primary group using the Agent Id;

1. Lift handset,
2. Dial '581' the ACD Agent Primary Login code,
3. Dial the ACD group number
4. Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the primary ACD Group;

1. Lift handset,
2. Dial ‘582’ the ACD Agent Primary Logout code.
3. Dial the ACD group number
4. Dial the Agent ID, the agent is logged out of the ACD Group.

To Login to a secondary group using the Agent Id;

1. Lift handset,
2. Dial ‘583’ the ACD Agent Secondary Login code,
3. Dial the ACD group number
4. Dial the Agent ID, the agent is logged into the ACD group.

To Logout of the secondary ACD Group;

1. Lift handset,
2. Dial ‘584’ the ACD Agent Secondary Logout code.
3. Dial the ACD group number ,
4. Dial the Agent ID, the agent is logged out of the ACD Group.

Conditions

1. The system outputs ACD events including Login/Logout messages when Discovery Manager Print is enabled.
2. The Agent Id can be any 4-digit number (0000-9999). iPECS does not verify the Agent Id, other than requiring that four digits be entered.
3. Any Agent Id may be used to login except at a Hot Desk station where the user Authorization code is required.

Programming

NUMBERING PLAN

1. ACD Agent Primary Login code (PGM 109-Btn 5)
2. ACD Agent Primary Logout code (PGM 109-Btn 6)
3. ACD Agent Secondary Login code (PGM 109-Btn 7)
4. ACD Agent Secondary Logout code (PGM 109-Btn 8)

STATION GROUPS

1. ACD Group (PGM 191)

SYSTEM

1. Discovery Manager Print (PGM 160-Btn 13)
2. Application Interface Msg (PGM 161-Btn 18)

Related Features

Station Groups
Hot Desk
Authorization Codes (Password)

Hardware

2.5.1.3 Agent Help Request

Description
Agents request assistance from a Supervisor using the ACD Help Code, default 574. Using iPECS Phones, agents with the **{HELP}** button can request assistance without interrupting an active conversation.

Operation
Agent iPECS Phone

*To assign **{HELP}** buttons:*

[PGM] + **{FLEX}** + '574' + ACD Group Number (optional) + **[SAVE]**

To request Supervisor assistance while on an active ACD call,

1. Press the **{HOLD}** button
2. Dial '574', the Agent Help code,
3. Dial the desired ACD Group number.

Or,

1. Press the **{HELP}** button.

SLT

1. Hook-flash.
2. Dial '574', the Agent Help code.
3. Dial the desired ACD Group number.

Conditions
1. Agents receive error tone to indicate there is no active Supervisor.
2. Using the **{HELP}** button during a call, no signals are sent on the active conversation and the connected party is unaware of the request for assistance.

Programming
NUMBERING PLAN
1. ACD Group Help Code (PGM 107-Btn 5)

STATION GROUPS
1. ACD Group (PGM 191)
2. ACD Supervisor (PGM 191-Btn 18)

Related Features
Station Groups
Supervisor Help Response

Hardware

2.5.1.4 Agent Queued Calls Display

Description

An Agent can view the queued call status for an ACD group when not on a call, in off-duty or wrap-up status. In addition, an active Agent can view the queued call status for an ACD group while on a call using **{Display Call Queue}** Flex button. The Call Queue display will show the ACD group number, the number of calls in queue and the longest queue time.

Operation

Agent iPECS Phone

*To assign a **{DISPLAY CALL QUEUE}** buttons;*

[PGM] + **{FLEX}** + '575' + ACD Group Number + **[SAVE]**

To display ACD Call Queue status when the Agent is idle and on-duty;

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial '575', the ACD Call Queue status code
3. Dial the ACD Group number. The display shows the Queued call status and will update as the queue status changes. If no calls are in queue, the phone returns to idle.
4. Hang-up to return to idle and normal display.

Or,

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **{DISPLAY CALL QUEUE}** button. The display shows the Queued call status and will update as the queue status changes. If no calls are in queue, the phone returns to idle.
3. Hang-up to return to idle and normal display.

To display ACD Call Queue status when the Agent is on a call;

1. Press the **{DISPLAY CALL QUEUE}** button, the displays shows the queued status for 2 seconds and then returns to the normal call display.

Conditions

1. The Agent cannot display queued call status while the phone is ringing.
2. If there are no queued calls to display, the phone returns to idle after providing error tone.

Programming

NUMBERING PLAN	1. Display ACD Call Queue Status (PGM 107-Btn 6)
STATION GROUPS	1. ACD Group (PGM 191)

Related Features

Station Groups

Hardware

iPECS Phone with Display

2.5.1.5 Agent Automatic Wrap-Up

Description

When an Agent completes an ACD group call, the Agent automatically enters into the Wrap-up state. In this state, an Agent will not receive ACD calls, allowing the Agent to complete paperwork, etc. The Agent remains in this automatic Wrap-Up state for the duration of the ACD group' Wrap-Up Timer. After the Wrap-Up Timer or by using 'Wrap-Up-End' feature, the Agent returns to available status and can receive ACD group calls.

Operation

Agent iPECS Phone

*To assign a **{WRAP-UP-END}** button;*

[PGM] + **{FLEX}** + '585' + **[SAVE]**

Activation;

Automatic when Agent completes an ACD group call

Deactivation;

1. Automatically after Wrap-Up Timer.

Or,

1. Dial '585' the Wrap-Up-End code.

Or,

1. Press **{WRAP-UP-END}** flexible button, before expiration of the Wrap-Up Timer.

Conditions

1. During Agent automatic Wrap-Up, the assigned **{WRAP-UP-END}** flexible button flashes. The button LED extinguishes when the Wrap-Up Timer expires or if the button is pressed, both return the Agent to available.

Programming

NUMBERING PLAN	1. ACD Agent Wrap-Up-End Code (PGM 109-Btn 9)
STATION GROUPS	1. ACD Group (PGM 191) 2. ACD Wrap-up Timer (PGM 191-Btn 9)

Related Features

Hardware

2.5.1.6 Agent On/Off Duty w/Reason Code

Description

Agents can control their status, On/Off-duty, and assign a Reason code for an Off-duty state. The system outputs the Reason code as part of the ACD Event message output. The Reason code may be any digit (0 ~ 9, * and #). With the exception of Reason code '0', when the Agent goes Off-duty manually, the Agent must return to the On-duty state manually.

If assigned as the Auto ACD DND code, using Reason code '0' activates the ACD DND Wrap-up timer. At expiration of the timer, the Agent is placed back On-duty. This provides the flexibility of an Agent activated Wrap-up time.

The Agent may assign an **{ACD ON/OFF DUTY}** button to for easy access. The ACD group number and the Reason code can be assigned for an **{ACD ON/OFF DUTY (GROUP) (REASON)}** button. By entering the Auto ACD DND code, the Agent will have an ACD DND button and, if assigned Reason code '0', the Agent will return to On-duty status after the ACD DND Wrap-up time.

Operation

Agent iPECS Phone

*To assign an **{ACD ON/OFF DUTY (GROUP)}** button;*

[PGM] + **{FLEX}** + '571' + (optional ACD group number) + (optional Reason Code) + **[SAVE]**

To toggle to Off-duty state from On-Duty,

1. Dial '571', Agent On/Off Duty Code.
2. Dial ACD group number
3. Dial Reason Code ('0' ~ '9', '*' or '#')

Or,

1. Press **{ACD ON/OFF DUTY (GROUP)}** button.
2. Dial Reason Code ('0' ~ '9', '*' or '#')

Or,

1. Press **{ACD ON/OFF DUTY (GROUP)(REASON)}** button.

To return to an On-duty state;

1. Dial '571', Agent On/Off Duty Code
2. Dial ACD group number

Or,

1. Press the **{ACD ON/OFF DUTY (GROUP)}**

Or,

1. Press the **{ACD ON/OFF DUTY (GROUP)(REASON)}** button.

*To assign an **{ACD ON/OFF DUTY IN ALL HUNT GROUP}** button;*

[PGM] + **{FLEX}** + the code for {Agent On/Off duty in all hunt group} + (optional Reason Code) + **[SAVE]**

The code for {Agent On/Off duty in all hunt group} is in PGM109-Btn 17.

To go to ON duty in all hunt group when the user is OFF duty in more one hunt group

1. Dial the code for {Agent On/Off duty in all hunt group}.

Or,

1. Press **{ACD ON/OFF DUTY IN ALL HUNT GROUP}** button.

Or,

1. Press **{ ACD ON/OFF DUTY IN ALL HUNT GROUP(REASON)}** button.

To go to OFF duty in all hunt group when the user is ON duty in all hunt group;

1. Dial the code for {Agent On/Off duty in all hunt group}.
2. Dial Reason Code (‘0’ ~ ‘9’, ‘*’ or ‘#’)

Or,

1. Press **{ACD ON/OFF DUTY IN ALL HUNT GROUP}** button.
2. Dial Reason Code (‘0’ ~ ‘9’, ‘*’ or ‘#’)

Or,

1. Press **{ ACD ON/OFF DUTY IN ALL HUNT GROUP(REASON)}** button.

SLT

To toggle to Off-duty state from On-Duty;

1. Lift handset or press **[SPEAKER]**,
2. Dial ‘571’, Agent On/Off Duty Code
3. Dial ACD group number
4. Dial Reason Code (‘0’ ~ ‘9’, ‘*’ or ‘#’)

To return to an On-duty state;

1. Lift handset or press **[SPEAKER]**,
2. Dial ‘571’, Agent On/Off Duty Code
3. Dial ACD group number

To go to ON duty in all hunt group when the user is OFF duty in more one hunt group

1. Lift handset or press **[SPEAKER]**,
2. Dial the code for {Agent On/Off duty in all hunt group}.

To go to OFF duty in all hunt group when the user is ON duty in all hunt group;

1. Lift handset or press **[SPEAKER]**,
2. Dial the code for {Agent On/Off duty in all hunt group}.
3. Dial Reason Code (‘0’ ~ ‘9’, ‘*’ or ‘#’)

Conditions

1. The system will output an Agent Availability message with the dialed Reason code when an Agent changes availability status. The message will include the button type “Wrap” if the ACD DND timer is activated.

2. The **{ACD ON/OFF-DUTY}** flex button will flash while the agent is Off-duty for any reason. The button LED extinguishes when the Agent returns to On-duty status.
3. While Off-duty, the supervisor's flex button LED assigned for the Agent's station flashes at the DND rate. The supervisor may call the station overriding the Off-duty status.
4. The **{ACD ON/OFF-DUTY (GROUP)(0)}** button overwrites other **{ACD DUTY (GROUP)}** buttons, only one **{ACD ON/OFF-DUTY (GROUP)** button can be assigned at a station.
5. The default value of the ACD DND Wrap-Up Timer is 10 seconds.

Programming

NUMBERING PLAN	1. Agent On/Off Duty code (PGM 107-Btn 2)
	2. Agent ON/OFF duty in all hunt group (PGM 109-Btn 17)
STATION	1. Auto ACD DND Reason code (PGM 113-Btn 15)
STATION GROUPS	1. ACD Group (PGM 191)
	2. ACD DND Wrap-up Timer (PGM 191-Btn 20)

Related Features

Agent Automatic Wrap-Up
Auto ACD DND Unavailable Service
Event Messages

Hardware

2.5.2 Announcements

Description

Each ACD group can provide announcements to incoming callers. Primary and secondary announcements are available with control timers. The first announcement is played after 1st control timer. The 2nd control timer determines the period between the first and second announcement. The second announcement can be replayed at defined intervals. If desired, the primary message can be defined to play in full to all callers as a 'guaranteed announcement' or only for calls that queue.

When enabled, a CIQ (Calls in Queue) announcement is played to callers that are queued to the ACD group. The CIQ announcement ("Your Call is number xx in Queue") is played to queued callers after the 1st announcement is played and again after each time the 2nd announcement is played. Internal callers with an iPECS Display Phone also receive the "You are xx in queue" display message.

Operation

System

ACD group announcements are played automatically based on the ACD group assignments.

System Attendant

To record a VMIM/VSF Announcement for an ACD Group;

1. Press the **[PGM]** button.
2. Dial '06', the VMIM/VSF Record code.
3. Dial VSF/VMIM sequence number to select a VSF/VMIM.
4. Dial the VMIM/VSF Announcement number (01-70)
5. Dial record language type (1~3), the current announcement is played followed by the "Press # to record" prompt.
6. Dial '#'.
7. After the beep-tone, record message.
8. Press the **[SAVE]** button to stop recording and save the message.

To delete a recording

1. Press the **[PGM]** button.
2. Dial '06', the VMIM/VSF Record code.
3. Dial VSF/VMIM sequence number to select a VSF/VMIM.
4. Dial the VMIM/VSF Announcement number (01-70)
5. Dial record language type (1~3), the current announcement is played followed by the "Press # to record" prompt.
6. Press the **[SPEED]** button during playback to erase message

Conditions

1. ACD group announcements are recorded in the VSF or VMIM. Up to seventy (70) Announcements are available for group announcement recordings. Announcements 71 and 72 are reserved for MOH and Multi Language selection announcements, respectively.
2. The CIQ Message must be recorded in the VMIM/VSF.
3. When equipped to support multiple languages, the Language selection announcement is played prior to a guaranteed announcement.
4. To define a guaranteed announcement, assign 0 seconds to the 1st announcement control timer.

Programming

STATION GROUPS

1. ACD Group (PGM 191)
2. ACD ICLID Routing (PGM 191-Btn 21)
3. CIQ Mention (PGM191-Btn 24-21)

Related Features

VMIM/VSF-Auto Attendant
ACD Caller Controlled ICLID Routing
Multiple Language Support

Hardware

2.5.3 Auto ACD DND Unavailable Service

Description

An Agent who does not answer a call offered from an ACD group during the ACD No-Answer timer is placed in a 'No response' unavailable status, called ACD DND. In this state, the Agent will not receive ACD calls from the group. The status is maintained until the agent manually returns to available status or, if the Auto ACD DND Reason code is set to '0', after the ACD DND Wrap timer the Agent returns to available status.

The Agent On/Off Duty code toggles the Agent duty status On and Off. The station may be assigned an **{ACD ON/OFF-DUTY (GROUP)}** button to set the active Agent duty status. The **{ACD ON/OFF-DUTY (GROUP)}** button LED will flash to indicate an unavailable status.

Operation

Agent iPECS Phone

*To assign a **{ACD ON/OFF-DUTY (GROUP)}** button;*

[PGM] + **{FLEX}** + '571' Agent On/Off Duty code + (optional ACD group number) + **[SAVE]**

To return to available status;

1. Dial '571', Agent On/Off Duty Code
2. Dial ACD group number

Or,

1. Press **{ACD ON/OFF-DUTY (GROUP)}**

SLT

To return to available status;

1. Lift handset or press [speaker],
2. Dial '571', Agent On/Off Duty Code
3. Dial ACD group number,

Conditions

1. The system will output an Agent Available message with a Reason code when an Agent changes availability status.
2. While Unavailable, the supervisor's flex button LED assigned for the station number of the agent will flash at the DND rate and the supervisor may call the station.

Programming

NUMBERING PLAN	1. Agent On/Off Duty Code (PGM107-Btn 2)
STATION	1. Auto ACD DND Reason code (PGM 113-Btn 15)
STATION GROUPS	1. ACD Group (PGM 191)
	2. ACD No Answer Timer (PGM 191-Btn 24-22)

Related Features

Agent On/Off Duty w/Reason Code

Hardware

2.5.4 Calls-In-Queue routing

Description

When a caller is queued to an ACD Group, various announcements may be played and music on hold may be sent to the caller. The caller may dial a digit at any time while queued to exit the queue, except during a Guaranteed Announcement. The dialed digit is compared to digits defined in the ACD Group CIQ Route Table. If a match is found, the call is routed to the defined destination (Station, Station Group, etc.). If a match is not found, external callers receive an error message and are placed back in queue; internal callers are simply placed back in queue.

Operation

Operation of this feature is automatic based on caller input

Conditions

1. ACD CIQ routing is not supported during or within 5 seconds of a Guaranteed Announcement. Digits dialed during a Guaranteed Announcement use Caller Controlled ICLID routing.
2. The call must be in queue for the ACD group, dialing during ring back tone is not recognized by the iPECS.

Programming

STATION GROUPS

1. ACD Group (PGM 191)
2. ACD CIQ Route Table (PGM 191-Btn 23)

Related Features

Station Groups
Announcements
ACD Caller Controlled ICLID Routing

Hardware

2.5.5 Calls-In-Queue Page Alert

Description

The administrator can establish CIQ (Calls-In-Queue) thresholds for each ACD Group so that an assigned VMIM/VSF announcement plays over a defined Page zone, if the number of calls in queue equals or exceeds a CIQ threshold. The message, which is recorded by an Attendant, plays

immediately or after a defined delay. The message repeats at assigned intervals until the number of Calls in Queue falls below the threshold. iPECS allows different treatment and VMIM/VSF announcements for up to three different Calls-In-Queue thresholds.

Operation

System

When the number of Calls-In-Queue changes, up or down the iPECS;

1. Checks CIQ thresholds,
2. Determines if threshold is reached,
3. Queues announcement for threshold (sets Delay timer),
4. Plays CIQ announcement.
5. Repeats announcement as defined.

System Attendant

To record a CIQ Announcement for an ACD Group;

1. Press the **[PGM]** button.
2. Dial '06', the Message Record code.
3. Dial VSF/VMIM sequence number to select a VSF/VMIM.
4. Dial the VMIM/VSF Announcement number (01-70)
5. Dial language selection (1~3), the current announcement is played followed by the "Press # to record" prompt.
6. Dial '#'.
7. After the beep-tone, record message.
8. Press the **[SAVE]** button to stop recording and save the message.

To delete a recording

1. Press the **[PGM]** button.
2. Dial '06', the Message Record code.
3. Dial VSF/VMIM sequence number to select a VSF/VMIM.
4. Dial the VMIM/VSF Announcement number (01-70)
5. Dial language selection (1~3), the current announcement is played followed by the "Press # to record" prompt.
6. Press the **[SPEED]** button during playback to erase message

Conditions

1. Thresholds are assigned as total Calls-In-Queue to activate a Page alert and are checked from lowest priority (CIQ #1) to highest priority (CIQ #3). Thus, the highest priority CIQ defined should have the largest threshold and the lowest priority CIQ should have the smallest threshold.
2. The VMIM and VSF have seventy (70) announcements employed for Auto Attendant, ACD groups, CCR, Page Alerts, etc.

Programming

STATION GROUPS

1. ACD Group (PGM 191)
2. CIQ #1 Threshold (PGM 191, Btn 24-6)
3. CIQ #1 Announcement Location (PGM 191-Btn 24-7)
4. CIQ #1 Page Zone (PGM 191-Btn 24-8)
5. CIQ #1 Announcement Delay Timer (PGM 191-Btn 24-9)
6. CIQ #1 Announcement Repeat Timer (PGM 191-Btn 24-10)
7. CIQ #2 Threshold (PGM 191-Btn 24-11)
8. CIQ #2 Announcement Location (PGM 191-Btn 24-12)
9. CIQ #2 Page Zone (PGM 191-Btn 24-13)
10. CIQ #2 Announcement Delay Timer (PGM 191-Btn 24-14)
11 CIQ #2 Announcement Repeat Timer (PGM 191-Btn 24-15)
12. CIQ #3 Threshold (PGM 191-Btn 24-16)
13. CIQ #3 Announcement Location (PGM 191-Btn 24-17)
14. CIQ #3 Page Zone (PGM 191-Btn 24-18)
15. CIQ #3 Announcement Delay Timer (PGM 191-Btn 24-19)
16 CIQ #3 Announcement Repeat Timer (PGM 191-Btn 24-20)
17 VSF Group Number (PGM 190)

Related Features

Station Groups
VMIM/VSF Integrated Auto Attd/Voice Mail
Internal/External & All Call Page

Hardware

2.5.6 Event Messages

Description

Each ACD group sends messages for group and agent events over the defined Serial or TCP port. These event messages are employed by 3rd party applications to provide enhanced Call Center functionality. Details of the event messages are provided in the iPECS 3rd Party Support Guide.

Operation

System

When enabled, Event messages are sent automatically

Conditions

Programming

STATION GROUPS
1. ACD Group (PGM 191)

SYSTEM
1. ACD Pack (PGM 175-Btn 9)
2. Discovery Manager Print (Output ACD Event Msg) (PGM 160-Btn 13)

Related Features

Station Groups

Hardware

2.5.7 Group Mail Box

Description

A mailbox can be associated with a Station Group. Calls to the group are assigned to overflow and can be otherwise re-routed to the Station Group Mail box. When routed to Voice Mail, messages are placed in the assigned Station Group Msg Wait Station mailbox.

Messages are retrieved in the same manner as normal voice messages employing the Station Group number as the station number and the defined Station Mailbox Password. Messages can be retrieved only if a **{GROUP MAILBOX}** Flex button is assigned to the station.

Operation

Agent iPECS Phone

*To assign a **{GROUP MAILBOX}** Flex button;*

[PGM] + **{FLEX}** + VM Group + Group Msg Wait Station + **[SAVE]**

To retrieve Station Group Voice Mail

1. Lift the handset or press the **[SPEAKER]** button
2. Press **{GROUP MAILBOX}** button to receive the “Password” prompt.
3. Dial the Group Mailbox password to receive the “Number of Messages” prompt.
4. Dial desired option code.
5. At completion of session, hang-up to return to idle.

Remote Caller

To leave a voice message

1. After Greeting and beep, leave the message.
2. Hang up to quit recording or dial ‘*’ for further options.

Conditions

1. The System Administrator or Supervisor may change the Group Mailbox password.

2. An external caller (CO/IP call) cannot access the Group Mailbox to retrieve messages.

Programming

STATION GROUPS
1. Group Attributes (PGM 191)
2. Mailbox Message Wait Station (PGM 191-Btn 24-2))
3. Mailbox Password (PGM 191-Btn 24-3))

Related Features

External Auto Attendant/Voice Mail
Station Groups
VMIM/VSF Voice Mail
Supervisors

Hardware

iPECS Phone

2.5.8 Group Name

Description

Each hunt group is assigned a name of up to 12 characters. The name is employed as the called party display for internal callers in the output of group statistics.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

Programming

STATION GROUPS
1. Station Group Attributes (PGM 191)
   Terminal/Cirular Group Name (PGM 191-Btn 20)
   ACD Group Name (PGM 191-Btn 22)
   Ring Group Name (PGM 191-Btn 18)
   VM Group Name (PGM 191-Btn 9)
   UCS Group Name (PGM 191-Btn 16)

Related Features

Station Groups

Hardware

2.5.9 Incoming Call Routing

Description

Incoming calls to an ACD group route directly to the station/agent that has been idle the longest (Uniform Call Distribution) or an ACD group announcement recorded by an Attendant. If all agents are busy, the call is queued to the group or routes to the Alternate Destination. The queued caller receives ACD group announcements or audio from the defined music source. Callers that remain in queue receive audio from the defined music source or ring-back tone and, after the defined Overflow time, route to an Overflow destination.

Agents can be assigned a priority from ‘0’ ~ ‘9’, ‘0’ being the lowest priority and ‘9’ the highest. When priorities are assigned, calls are routed to the highest priority Agent that has been idle the longest.

If there are no Agents active in the group when a call arrives, the call will receive the “No Member” treatment. Intercom calls receive re-order and external callers are routed to an Attendant.

Operation

System

Routing of ACD group calls is automatic based on the ACD group assignments.

Conditions

1. The Alternate ACD Destination can be a station or a group but cannot be a member of the ACD group.
2. The Overflow destination may be any station or group including the ACD group VMIM/VSF Mailbox but cannot be a member of the overflowing ACD group.
3. If an agent becomes available during an announcement, except a guaranteed announcement, the call is passed immediately to the agent.

Programming

STATION GROUPS

1. ACD Group (PGM 191)
2. Overflow Destination (PGM191-Btn 7)
3. Overflow Time (PGM 191-Btn 8)
4. Music Source (PGM 191-Btn 11)
5. Alternate Destination (PGM 191-Btn 13)
6. Report No Members (PGM 191-Btn 10)
7. Agent Priority (PGM 191-Btn 19))

Related Features

Automatic Call Distribution

Hardware

2.5.10 ACD Caller Controlled ICLID Routing

Description

With Caller Controlled ICLID enabled (ICLID Usage) for the ACD Group, at any time during or within 5 seconds after a Guaranteed Announcement, the system will monitor for digits dialed by the caller. Received digits are compared to the ICLID Table entries. If a match is found, the call routes to the destination defined in the ICLID Ring Assignment Table. Available destinations are a Voice Mail announcement, system speed dial, a station or a station group. If a match is not found, the call returns to the queue and is handled as defined for the ACD group.

Operation

System

ICLID Routing of ACD group calls is automatic based on the ACD group assignments.

Conditions

1. The conditions of VMIM/VSF AA Announcements and ICLID Call Routing apply.
2. Only the caller-entered digits are employed for ACD ICLID routing.
3. The ICLID Table routing is only followed if the 1st ACD announcement is Guaranteed, ICLID Usage is enabled and the digits are entered during the 1st Announcement.
4. The caller may correct an entry by dialing * and dialing the correct entry.

Programming

STATION GROUPS
1. ACD Group (PGM 191)
2. ICLID Usage (PGM191-Btn 21)

TABLES
1. ICLID Table (PGM 203)
2. ICLID Ring Assignment (204)

Related Features

Automatic Call Distribution
ICLID Call Routing
VMIM/VSF-Auto Attendant

Hardware

2.5.11 Supervisors

2.5.11.1 Supervisor Login/Logout

Description

Each ACD group is assigned up to five (5) Supervisors to monitor and control real-time status of the ACD group and Agents assigned to the group. A Supervisor can be assigned to multiple groups. The Supervisor login/logout feature provides a means for a supervisor to log into one or more ACD groups and monitor calls.

Each Supervisor has a unique 4-digit Agent Id used to login and logout of active ACD group participation. Supervisors can login from any station in the system but only at one station at a time. If a Supervisor logs into a group from a station while logged in at another station, the active login is terminated and the new login is honored.

Operation

Supervisor iPECS Phone

To login;

1. Dial '572' the ACD Supervisor Login code or press the Flex button.
2. Dial the ACD group number
3. Dial supervisor Id code ('0000'-'9999')

To Logout;

1. Dial '573' the ACD Supervisor Logout code or press the Logout Flex button.
2. Dial the ACD group number
3. Dial supervisor Id code ('0000'-'9999'')

Conditions

1. The system will output a Login/Logout message when a Supervisor logs-in or out.
2. The Supervisor Id can be any 4-digit number ('0000'-'9999').
3. iPECS system does not verify the Supervisor Id codes, other than requiring four digits are entered.
4. Any Agent Id may be used to login through stations not assigned as a Hot Desk.
5. The ACD Supervisor can have an ACD Group Status button for each group to indicate Group activity.

Programming

NUMBERING PLAN	1. Supervisor Login code (PGM 107-Btn 3) 2. Supervisor Logout code (PGM 107-Btn 4)
STATION GROUPS	1. ACD Groups (PGM191) 2. ACD Group Supervisor (PGM 191-Btn 18)

Related Features

Automatic Call Distribution
Hot Desk

Hardware

iPECS Phone

2.5.11.2 Supervisor Help Response

Description

An Agent may request assistance from a Supervisor using the ACD Group Help Code, default 574 or **{HELP}** button. The Supervisor is then notified of the Help request in the iPECS Phone display and a flashing **{HELP RESPONSE}** button. The Supervisor can respond using the Help Response button, which is required, and be connected to the Agent's active call with the microphone muted. The Supervisor hears the Agent and connected party and may join the conversation by pressing the **[MUTE]** button to activate the microphone.

Operation

Agent iPECS Phone

To request Supervisor assistance,

1. Press the **{HOLD}** button
2. Dial '574', the Agent Help code,
3. Dial the desired ACD Group number.

Or,

1. Press the **{HELP}** button.

SLT

1. Hook-flash and dial '574', the Agent Help code
2. Dial the desired ACD Group number.

Supervisor iPECS Phone

*To assign a **{HELP RESPONSE}** button*

[PGM] + **{FLEX}** + '574' + "Group Number" + **[SAVE]**

To respond to a Help request, at the Supervisor Phone,

1. Lift the handset or press the [speaker] button,
2. Press the flashing **{HELP RESPONSE}** button, the Supervisor is connected to the active Agent call with the microphone muted, the **[MUTE]** button LED is On.

To converse with the Agent and connected party

1. Press the illuminated **[MUTE]** button.

Conditions

1. The Agent Help request is sent to the lowest numbered available Supervisor station for the associated ACD Group.
2. If no Supervisor is available, the Agent receives error tone.
3. A Warning Tone, if enabled, will be sent to the Agent as the Supervisor is connected to the conversation.
4. Only one active Help request is allowed at a time, a second help request will cancel any active help request.

Programming

NUMBERING PLAN
1. ACD Group Help Code (PGM 107-Btn 5)

STATION GROUPS
1. ACD Groups (PGM 191)
2. ACD Group Supervisor (PGM 191-Btn 18)
3. ACD Warning Tone (PGM 191-Btn 12)

Related Features

Automatic Call Distribution
Agent Help Request

Hardware

iPECS Phone

2.5.11.3 Agent Call Monitor

Description

Agent Call Monitor permits an active Supervisor to monitor an Agent's call in progress for training purposes or to assist the Agent. When used, the Supervisor is connected to the call with the microphone muted, the **[MUTE]** button LED is On. If ACD Warning Tone is enabled, a Warning Tone is provided to the Agent's call indicating the Supervisor has entered the call. The Supervisor hears the Agent and connected party and may join the conversation by pressing the **[MUTE]** button to activate the microphone.

The Supervisor station must be assigned an **{AGENT MONITOR}** button to activate the Agent Call Monitor feature.

Operation

Supervisor iPECS Phone

To assign an {AGENT MONITOR} button

[PGM] + **{FLEX}** + '577' + "Group Number" + **[SAVE]**

To enter an active Agent call from the Supervisor phone

1. Lift the handset or press the **[SPEAKER]** button.
2. Call the desired Agent and receive busy tone,
3. Press the **{AGENT MONITOR}** button, the Supervisor is connected to the active Agent call with the microphone muted, the **[MUTE]** button LED is On.

To converse with the Agent and connected party

1. Press the illuminated **[MUTE]** button.

Conditions

1. The Supervisor must be active and logged into the Agent's ACD group to monitor calls in progress.

Programming

NUMBERING PLAN
1. Supervisor Monitor Code (PGM 107-Btn 8)

STATION GROUPS
1. ACD Groups (PGM 191)
2. ACD Group Supervisor (PGM 191-Btn 18)
3. ACD Warning Tone (PGM 191-Btn 12)

Related Features

Station Groups

Hardware

iPECS Phone

2.5.11.4 Group Status Display

Description

An ACD Supervisor may view the status of each of the ACD groups to which the Supervisor is assigned. The display will give the status of the active ACD group at the time of the request. The display indicates the following statistics:

Number of calls in Queue
Wait time for the call in queue longest and
Agents in service and available

When a call is queued to the group, the ACD Supervisor is notified, in real-time, by the LED of the **{GROUP STATUS}** button. The **{GROUP STATUS}** button allows access to the ACD Group status display or the Group Status code may be used to display the status information.

Operation

Supervisor iPECS Phone

*To assign a **{GROUP STATUS}** button at the Supervisor*

[PGM] + **{FLEX}** + '576' + "Group Number" + **[SAVE]**

*To display the Group status when the **{GROUP STATUS}** button is flashing*

1. Lift the handset or press the **[SPEAKER]** button,
2. Dial '576', the Group Status display code,
3. Dial '1' to see the Group Status information,

Or,

1. Press the flashing **{GROUP STATUS}** button,
2. Dial '1' to see the Group Status display.

Conditions

1. The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.
2. The **{GROUP STATUS}** button will flash when the Calls-in-Queue exceed the assigned Supervisor Call Count and/or a call has been queued for longer than the Supervisor Queued Call Timer.

Programming

NUMBERING PLAN
1. ACD Group Status Code (PGM 107-Btn 7)

STATION GROUPS
1. ACD Group (PGM 191)
2. ACD Group Supervisor (PGM 191-Btn 18)
3. Supervisor Timer (PGM 191-Btn 14)
4. Supervisor Queued Call Count (PGM191-Btn 15)

Related Features

Station Groups

Hardware

iPECS Phone

2.5.11.5 Group Parameter Control

Description

An ACD Group Supervisor can adjust several of the ACD Group routing parameters, in real-time. Adjustable parameters are:

- Overflow Destination
- Overflow Time
- Wrap-up Time
- ACD Voice Mailbox Password

Operation

Supervisor iPECS Phone

*To assign a **{GROUP STATUS}** button at the Supervisor*

[PGM] + **{FLEX}** + '576' + "Group Number" + **[SAVE]**

To use the Supervisor Status Menu

1. Lift the handset or press the **[SPEAKER]** button,
2. Press the **{GROUP STATUS}** button,

[1] ACD STATUS [2] ACD DATABASE [3] ACD DUTY [#] ACD PRINT

3. Dial ACD Dbase Code '2',
4. Select database item, scrolling with the **[VOL UP]**/**[VOL DWN]** button;
   Overflow Destination- station (Net station) (1)/group (2)/system speed dial (3)
   Overflow Time- xxx seconds
   Wrap-Up Time- xxx seconds
   Password- up to 12 digits
5. Enter new data,
6. Press the **[SAVE]** button.

Conditions

1. The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

Programming

NUMBERING PLAN	1. ACD Group Status Code (PGM 107-Btn 7)
STATION GROUPS	1. ACD Group (PGM 191)
	2. ACD Group Supervisor (PGM 191-Btn 18)

Related Features

Station Groups

Hardware

iPECS Phone

2.5.11.6 Agent Status Control

Description
An ACD Supervisor can view and control the status of Agents in the group.

Operation
Supervisor iPECS Phone

To assign a ***{GROUP STATUS}*** *button at the Supervisor*

[PGM] + **{FLEX}** + '576' + "Group Number" + **[SAVE]**

To view/control Agent Status

1. Lift the handset or press the **[SPEAKER]** button,
2. Press the **{GROUP STATUS}** button,

[1] ACD STATUS [2] ACD DATABASE [3] ACD DUTY [#] ACD PRINT

3. Dial '3', Agent Code, the LCD menu will update:

[1] DUTY STATUS [2] DUTY ON/OFF [#] DUTY PRINT

To view Agent Status

1. Dial '1', Agent Status Code,

ACD STATUS : XXX TOTAL CALLS : XXX

2. Press **[VOL UP]** or **[VOL DWN]** for:
   - Number of ACD calls served
   - Number of unanswered ACD Calls
   - Average ring time before answer
   - Average ACD call service time after answer.
3. . Press the '*' key for a next agent selection,

Or,

To control Agent duty status

1. Dial '3', ACD Agent Duty Code,
2. Dial '2' for Duty ON/OFF,
3. Dial '0' or '1', ('0': Off Duty, '1': On Duty).

Conditions
1. The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

Programming

NUMBERING PLAN
1. ACD Group Status Code (PGM 107-Btn 7)

STATION GROUPS
1. ACD Group (PGM 191)
2. ACD Group Supervisor (PGM 191-Btn 18)

Related Features

Station Groups

Hardware

iPECS Phone

2.5.12 ACD Statistics Report

ACD reports can be requested by the Supervisor and can be programmed for periodic output over the SMDR port or selected TCP port. The system will provide reports for the ACD Group and Agent statistics as follows:

<u>ACD Group Statistics Report</u>
Group Number
Time stamp
Total calls
Number of unanswered calls
Average queue time
Longest queue time
Total number of calls placed in queue
Number of times calls experience all agents busy
Total time all agents were busy
Average ring time before answer
Average service time after answer

<u>Agent Statistics Report</u>
Group Number
Agent Number
- Number of ACD calls served
- Number of unanswered ACD Calls
- Average ring time before answer
- Average ACD call service time after answer.

Operation

Supervisor iPECS Phone

*<u>To assign a **{GROUP STATUS}** button at the Supervisor</u>*

[PGM] + **{FLEX}** + '576' + "Group Number" + **[SAVE]**

To output ACD Statistics Report

1. Lift the handset or press the **[SPEAKER]** button,
2. Press the **{GROUP STATUS}** button,

1STATUS 2DBASE 3AGENT #PRINT

3. Dial '#', ACD Group Statistics Reporting Code,
4. Press the **[MUTE]** button to initialize the ACD database after printing; this eliminates overlap of future reports.

To output the Agent Statistics Report

1. Lift the handset or press the **[SPEAKER]** button,
2. Press the **{GROUP STATUS}** button,

1STATUS 2DBASE 3AGENT #PRINT

3. Dial '3', Agent Code, the LCD menu will update:

1STATUS/2ON/OFF)/#PRINT

4. Dial '#', Agent Statistics Reporting Code,
5. Press the **[MUTE]** button to initialize the ACD database after printing; this eliminates overlap of future reports.

Conditions

1. The ACD status can be printed periodically. The period is assigned in Admin Programming. The ACD records contain information for both ACD agents and ACD group.
2. The Supervisor must be assigned as a Supervisor for the ACD Group and be logged into the group to access the Group Status display information.

Programming

NUMBERING PLAN
1. ACD Group Status Code (PGM 107-Btn 7)

STATION GROUPS
1. ACD Group (PGM 191)
2. ACD Group Supervisor (PGM 191-Btn 18)

Related Features

Station Groups

Hardware

2.6 Automatic Pause Insertion

Description

In addition to a manually entered Pause, the system will automatically pause dialing to allow for potential connection delays. The pause will be inserted when any of the following occur:

- After a Flash is encountered in a Speed Dial number,
- After a PABX access code is encountered in a Speed Dial or redial number,
- After a Pulse to Tone Switchover is encountered in a Speed Dial or Redial number.
- When a connect message is received on an ISDN Line.

Operation

System

The system automatically pauses dialing after an appropriate event.

Conditions

1. An automatically inserted pause is not counted as a digit in a Speed Dial number.
2. The LCD of the iPECS Phone will show a "P" when a pause is encountered. This indication is not shown if the system inserts the Pause automatically.

Programming

SYSTEM 1. Pause Timer (PGM 181-Btn 10)

Related Features

Station Speed Dial
System Speed Dial
Auto Called Number Redial (ACNR)
LNR (Last Number Redial)
Dial Pulse to Tone Switchover

Hardware

2.7 Automatic Privacy

Description

Privacy is insured on all communications in the system. If desired, the customer may elect to disable the Automatic Privacy feature, allowing another station to join in an existing external conversation uninvited. In such a case, a conference is established.

Operation

iPECS Phone

To intrude into a call when Privacy is disabled

1. Press a busy (lit steady) individual **{CO}**/**{IP}** access button, user connected to the call with existing internal station user.

Conditions

1. With Automatic Privacy disabled, privacy is still assured on all intercom and conference calls.
2. To override privacy, Privacy must be disabled and the intruding station must have Override enabled as well as a direct appearance for the desired **{CO}/{IP}** line.
3. Only one station can intrude on an active call.
4. An intrusion tone can be provided to the call indicating another station has accessed the line.
5. If either internal party presses another **{CO}**/**{IP}**, a **{DSS}**, **{PAGE}**, **[CONF]** or other conflicting button, the party is removed from the "Conference" and must press the **{CO}**/**{IP}** button again to reenter the conversation.

Programming

STATION
1. Override Privilege (PGM 113-Btn 4)

SYSTEM
1. Privacy (PGM 161-Btn 3)
2. Privacy Warning Tone (PGM 161-Btn 4)

Related Features

Multi-Party Voice Conference
Station Flexible Buttons

Hardware

iPECS Phone

2.8 Auto Service Mode Control

Description
The service mode defines different ring assignments, COS and answering privileges for the system. The service mode can be controlled automatically through definitions in the Auto Ring Mode Selection Table, which defines the time of day for the Day, Night and Timed shift modes. The Attendant may change the system mode selection from automatic to manual.

Operation
System

Operation of this feature is automatic.

Conditions

Programming
STATION
1. Station COS (PGM 116)

CO/IP
1. CO Ring Assignment (PGM 144)
2. CO COS (PGM 141-Btn 2)

SYSTEM
1. DISA COS (PGM 166)
2. External Control Contact (PGM 168)
3. PBX Trunk Access Codes (PGM 172)
4. LD Digit Count (PGM 177-Btn 3)
5. LD Digit Code (PGM 177-Btn 19)

TABLES
1. Toll Restriction (PGM 224)
2. Authorization Codes Table (PGM 227)
3. Auto Ring Mode Selection Table (PGM 233)

Related Features
Off-Hook Signaling
Authorization Codes (Password)
Direct Inward System Access (DISA)
Day/Night/Timed/Scenario Ring Mode
System Clock Set
CO/IP Ring Assignment
LBC (Loud Bell Control)
Dialing Restrictions

Hardware

2.9 AUTOMATIC SYSTEM DAYLIGHT SAVINGS TIME

Description
The system can automatically adjust for Daylight Saving Time (DST). When DST is enabled, the system will adjust the system time forward one (1) hour at the DST Start time and back one (1) hour at the DST End time. The system time is sent for display to all devices and terminals and is the basis of the system various time-based features (Walking COS, Wake-up Alarm, etc.).

Operation
System

Operation of this feature is automatic.

Conditions
1. The DST Start and End times are set through the Web Admin interface only.
2. The interval between the DST Start and End times must be at least 7 days.

Programming
SYSTEM

1. System Time (PGM 178-Btn 1)
2. System Date (PGM 178-Btn 2)
3. DST Enable (PGM 178 – Btn 3)
3. DST Start & End Time (Web only)

Related Features
Auto Service Mode Control
Automatic System Time Synchronization
System Clock Set

Hardware

2.10 AUTOMATIC SYSTEM TIME SYNCHRONIZATION

Description
When enabled, the system automatically determines and sets the time of day employing Network Time Protocol (NTP) or ISDN time messages. When using NTP, the system, at ten (10) minute intervals, requests the time from the specified NTP time server and receives GMT time. This feature allows the System Time to synchronize with the NTP time server automatically. If the time deviates more then two (2) seconds, the system clock is adjusted to match the NTP server.

When using ISDN, the system receives the time of day in ISDN messages and automatically adjusts the time if the time in the system deviates from the ISDN time.

Operation

System

Operation of this feature is automatic.

Conditions

1. NTP packets are expected over UDP port 123. Assure this port is open and available.
2. A secondary NTP server address can be defined should the first server not respond.
3. The system adjusts for the local time zone assigned in the system as the Standard System Time as well as Daylight Savings Time (DST), if set.

Programming

SYSTEM

1. Network Time/Date (PGM 161 – Btn 12)
2. NTP Active (PGM 195 – Btn 1)
2. NTP Sever address (Web only)
3. Std system time, local Time Zone (Web only)

Related Features

Auto Service Mode Control
Automatic System Daylight Savings Time
System Clock Set

Hardware

2.11 BATTERY BACK-UP, MEMORY

Description

The system database is protected from power-loss by a long life (10-year) lithium dry cell battery. Should local power fail, the battery will maintain the system memory and proper operation of the system clock.

Operation

System

Operation of this feature is automatic.

Conditions

1. The Initialization switch must be in the OFF position to enable Memory Battery Back-up. Otherwise, should power fail, the system will initialize the database on power-up. Refer to the iPECS Hardware Description and Installation manual section 4.4.2.
2. The Lithium battery is not field-replaceable.

Programming

Related Features

Hardware

2.12 CALL FORWARD

Description

Users may have selected incoming calls re-routed to other stations (local or networked), station groups, the VMIM/VSF, or over a system CO/IP line (Off Net).

The user selects the type and condition under which calls will forward by entering a Call Forward code as follows:

Code 0:	Remote Call Forward, forwards all calls to the station, except recalls, activated from a remote station, Call Forward, Follow-me.
Code 1:	Unconditional, all calls to the station, except recalls, are forwarded internally or externally immediately upon receipt.
Code 2:	Busy, if the station is busy, forwards all calls, except recalls, to the selected station.
Code 3:	No Answer, forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer.
Code 4:	Busy/No Answer, forwards calls if the selected station is busy or does not answer within the No Answer timer.
Code 5:	Attendant Off-Premise, forwards incoming CO/IP calls to an outside number.
Code 6:	Off-Net Unconditional, all calls to the station, except recalls, are forwarded internally or externally, only SLT.
Code 7:	Off Net Busy, forwards all calls, except recalls, to the selected station when station is busy, only SLT.
Code 8:	Off Net No Answer, forwards all calls, except recalls, to the selected station when the station does not answer within the No Answer timer, only SLT.
Code 9:	Off Net Busy/No Answer, forwards calls if the selected station is busy or does not answer within the No Answer timer, only SLT.

Operation

iPECS Phone

To activate Call Forward, Unconditional or Busy/No Answer

1. Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2. Press the **[FWD]** button
3. Dial desired Call Forward code ('1'~'4').
4. Dial the station or station group to receive calls.
   or
   Dial CO access code (0, 8xx, 88xx) and desired external phone number.
   Press [SAVE] button to save.
   or
   Press **[SPEED]** button
   Dial desired bin number.
5. Replace the handset, return to idle.

To activate Call Forward, Remote (Follow-me)

1. Lift the handset or press **[SPEAKER]** button to receive dial-tone
2. Press the **[FWD]** button,
3. Dial Call Forward code '0',
4. Dial the station's Authorization Code (Station number + password),
5. Dial Forward condition ('1'~'4'),
6. Dial the destination station or station group,
   or
   Dial CO access code (0, 8xx, 88xx) and desired external phone number.
   Press [SAVE] button to save.
   or
   Press **[SPEED]** button
   Dial desired bin number.
7. Replace the handset, return to idle.

To deactivate Call forward

1. Press flashing **[FWD]** button, Call Forward will deactivate and the **[FWD]** button LED is off.

SLT

To activate Call Forward, Unconditional, Busy/No-Answer to an internal number

1. Lift the handset to receive dial tone.
2. Dial Call Forward feature dial code '554'
3. Dial desired Call Forward code ('1'~'4').
4. Dial the destination station, station group or dial CO access code (0, 8xx, 88xx) and desired external phone number.
5. Momentarily press the hook-switch, receive confirmation tone.
6. Replace the handset, return to idle.

To activate Call Forward to a Speed number

1. Lift the handset to receive dial tone.

2. Dial Call Forward feature access dial code ‘554’.
3. Dial Call Forward code (‘6’~’9’),
4. Dial Speed Dial bin number.
5. Replace handset to return to idle.

To activate Call Forward, Remote (Follow-me)

1. Lift the handset.
2. Dial the Call Forward feature code ‘554’.
3. Dial Remote Forward code ‘0’
4. Dial the station’s Authorization Code (Station + Password),
5. Dial Forward condition (‘1’~’4’)
6. Dial the destination station, station group .or dial CO access code (0, 8xx, 88xx) and desired external phone number.
7. Momentarily press the hook-switch, receive confirmation tone.
8. Replace handset return to idle.

To deactivate the Call forward

1. Lift the handset, receive stutter dial-tone,
2. Dial Call Forward feature access dial code ‘554’,
3. Dial ‘#’ to cancel Call Forward.

Conditions

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. A station, denied the use of Call Forward, will receive error tone in response to attempts to activate Call Forward.
3. A forwarded intercom call will signal the receiving station in the Tone Signaling mode, regardless of the Intercom Signaling Mode at the station.
4. Calls cannot be forwarded to a station in DND and, if attempted, error tone is returned.
5. Attempting activation of Call Forward will automatically deactivate any activate Display Text Message. Active Call Back or Queue requests are not canceled.
6. When Call Forward is active, a station can make outgoing calls (internal or external) but cannot activate a Call back or Queue request.
7. Private lines can be forwarded if the forwarding and receiving station have a direct appearance **{CO}** button for the Private line or if calls are forward to a VSF, VMIM or Voice Mail group.
8. For CO/IP calls, manually activated Call Forward will override any Preset Call Forward assigned for the station or CO/IP line.
9. Users may forward calls to the VMIM/VSF without recording a greeting, in which case, calls are still forwarded to the VMIM/VSF but callers receive the "No announcement recorded" message before they can leave a voice mail.
10. Call Forward status is maintained in the system’s non-volatile memory for protection from power outage.
11. A station in a Station Hunt Group (ACD, Circular or Terminal) can be assigned to receive incoming hunt calls, overriding any Call Forward. That is, the system either recognizes the

Forward condition and bypasses hunt calls around the station or routes hunt calls to the station based on the system database (Member Forward).

12. Off-Net Call Forward of incoming CO/IP calls is essentially an automated DISA call, which will establish an Unsupervised Conference. Such calls are subject to the conditions of a DISA call and Unsupervised Conference and may require entry of an Authorization Code.
13. Off-Net forward calls are not answered until the system completes dialing of the external call. The call, internal or external, is then connected to the Off-Premise call.
14. If the Speed Dial number used in Off-Net Call Forward contains a Flash, only digits prior to the Flash are dialed.
15. An unlimited number of stations may be set-up in a Call Forward chain, forwarding calls from one station to the next. However, a station cannot forward calls to a station already a part of the chain.
16. Calls to a Call Forward chain will progress as appropriate through the chain to the last station. If the last station enters DND, CO/IP calls revert to the previous station while intercom calls receive DND tone.
17. No Answer forward employs the Station No Answer Forward Timer unless it is set to zero in which case the System No Answer Timer is used.

Programming

STATION	1. Call Forward (PGM 111-Btn 2)
	2. Station Call Forward No Answer Timer (PGM 123-Btn 1)
SYSTEM	1. System Call Forward No Answer Timer (PGM 181-Btn 1)
STATION GROUPS	1. Member Forward (PGM 191-Btn 14-Circ/Term, 24-23-UCD, or 12-Ring)

Related Features

Call Forward, Attendant
Authorization Codes (Password)
DND (Do Not Disturb)
DND - One Time DND
Direct Inward System Access (DISA)
Unsupervised Conference
Dialing Restrictions
Station Groups
Station Speed Dial
System Speed Dial
Intercom Signaling Mode
Call Forward, Preset

Hardware

2.13 CALL FORWARD, PRESET

Description

With Call Forward, Preset, calls to a station forward to a pre-determined destination assigned in the system database. Preset Station Call Forward can define separate treatment of CO/IP calls and intercom calls. In addition, separate busy and no-answer treatments are defined. Treatments available are:

- Unconditional, all calls immediately forward,
- Internal Busy, Intercom calls that encounter busy, forward immediately.
- Internal No-Answer, Intercom calls, which are not answered in the No-Answer time, forward. Note calls to a busy station also forward after the No-Answer time.
- External Busy, external calls that encounter busy forward immediately.
- External No-Answer, external calls, not answered in the No-Answer time, forward. Note calls to a busy station also forward after the No-Answer time.

In addition, calls can be directly forward to the users Voice Mail box using Call Forward, Preset.

Operation

System

Operation of Preset Call Forward is automatic.

Conditions

1. A station receiving a forwarded call can transfer the call to the forwarding station.
2. Calls cannot be forwarded to a station in DND and, if attempted, error tone is returned.
3. Manual Forward has higher priority than Preset Forward and overrides any Preset Forward setting.
4. Calls to a Preset Call Forward chain will progress as appropriate through the chain to the last station. If a station in manual Call Forward or DND is encountered, it is bypassed and the next station in the chain is signaled. If the last station has entered DND, CO/IP calls revert to the previous station, signaling until answered or abandoned.
5. Internal Busy or No Answer will only operate when the internal call encounters a busy state or no answer, respectively. External Busy or External No Answer will only operate when the external call encounters a busy state or no answer, respectively.
6. Preset call forward status is not shown in the station’s LCD display.
7. A station in a Station Hunt Group (ACD, Circular or Terminal) can be assigned to receive incoming hunt calls, overriding any Call Forward. That is, either the system recognizes the Forward condition and bypass hunt calls around the station or routes hunt calls to the station based on the system database.
8. No Answer forward employs the Station No Answer Forward Timer unless it is set to zero in which case the System No Answer Timer is used.

Programming

STATION
1. Preset Call Forward (PGM 120)
2. Station Call Forward No Answer Timer (PGM 123-Btn 1)
3. Direct Transfer Mail Box Destination (PGM 120-Btn 6)

STATION GROUPS
1. Member Forward (PGM 191 Btn-14-Circ/Term, 24-23-UCD, or 12-Ring)

SYSTEM
1. System Call Forward No Answer Timer (PGM 181-Btn 1)

Related Features

Call Forward
Off-Hook Signaling
External Auto Attendant/Voice Mail
VMIM/VSF Integrated Auto Attd/Voice Mail
DND (Do Not Disturb)
DND - One Time DND
Private Line

Hardware

2.14 CALL PARK

Description

A user may place an active CO/IP call in a special holding location (Park Orbit) for easy access from any station in the system.

Operation

iPECS Phone

To park an active external call

1. Press the **[TRANS]** button.
2. Dial the Park Orbit.
3. Return to idle.

To retrieve a parked call

1. Lift the handset or press the **[SPEAKER]** button,
2. Dial the Park Orbit.

SLT

To park an active external call

1. Momentarily press the hook-switch.

2. Dial the Park Orbit.
3. Return to idle.

To retrieve a parked call

1. Lift the handset.
2. Dial the Park Orbit.

Conditions

1. If the selected Park Orbit returns a busy signal, the user may simply dial another Park Orbit without the need to disconnect.
2. Intercom calls cannot be placed in a Park Orbit.
3. A Parked call will recall to the station that parked the call should the Call Park Timer expire. The normal Hold Recall process is then initiated.
4. A Parked call will indicate busy at all appearances.

Programming

SYSTEM 1. Call Park Recall Timer (PGM 180-Btn 2)

Related Features

Hold Recall
Attendant Recall

Hardware

2.15 CALL PICK-UP

2.15.1 Directed Call Pick-Up

Description

A station may answer ("Pick-Up") incoming and transferred intercom, CO and IP calls ringing at another station. All ringing calls are subject to Directed Call Pick-up except Private Line and queue Callbacks.

iPECS phone users may assign a Flex button as a **{DIRECTED CALL PICK-UP}** button.

Operation

iPECS Phone

To assign a {DIRECTED CALL PICK-UP} button

[PGM] + **{FLEX}** + '*7' + **[SAVE]**

To Pick-up a call ringing at another station

1. Lift the handset or press **[SPEAKER]**.
2. Dial '*7', the Directed Call Pick-up feature code.
3. Dial the ringing station's intercom number.

Or,

1. Lift the handset or press **[SPEAKER]**.
2. Press the **{DIRECTED CALL PICK-UP}** button.
3. Dial the ringing station's intercom number.

SLT

To Pick-up a call ringing at another station

1. Lift the handset
2. Dial '*7', the Directed Call Pick-up code.
3. Dial the ringing station's number.

Conditions

1. To pick-up a CO/IP call, the station must have an idle appearance button available.
2. When several calls are ringing at a station simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173).
3. Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
4. Handsfree announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.

Programming

SYSTEM 1. Ringing Line Preference Priority (PGM 173)

Related Features

Intercom Signaling Mode
Ringing Line Preference
Group Call Pick-Up
Private Line

Hardware

2.15.2 Group Call Pick-Up

Description
A station can answer (“Pick-Up”) incoming and transferred intercom, CO and IP calls ringing at another station in the same station group. All ringing calls, except Private Line and Queue Callbacks, are subject to Pick-up by other stations in the same group.

iPECS phone users may assign a Flex button as a **{GROUP CALL PICK-UP}** button.

Operation
iPECS Phone

*To assign a **{GROUP CALL PICK-UP}** button*

[PGM] + **{FLEX}** + ‘**’ + **[SAVE]**

To Pick-up a call ringing at another station
1. Lift the handset or press **[SPEAKER]**.
2. Dial ‘**’, the Group Call Pick-up feature code.

Or,
1. Press the **{GROUP CALL PICK-UP}** button.

SLT

To Pick-up a call ringing at another station
1. Lift the handset
2. Dial ‘**’, the Group Call Pick-up code.

Conditions
1. To pick-up a CO/IP call, the station must have an idle appearance button available.
2. When several calls are ringing simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173).
3. Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
4. Handsfree announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.
5. When a station belongs to multiple groups, calls to the group with the lowest number are answered first.

Programming

STATION GROUPS	1. Station Group (PGM 190-Btn 1)
SYSTEM	1. Ringing Line Preference Priority (PGM 173)

Related Features

Intercom Signaling Mode
Ringing Line Preference
Directed Call Pick-Up
Station Groups
Private Line

Hardware

2.15.3 Pick Up Group Call Pick-Up

Description

A station can answer ("Pick-Up") incoming and transferred intercom, CO and IP calls ringing at another station in the same pick up group. All ringing calls, except Private Line and Queue Callbacks, are subject to Pick-up by other stations in the same group.

iPECS phone users may assign a Flex button as a **{PICK UP GROUP CALL PICK-UP}** button.

Operation

iPECS Phone

*To assign a **{PICK UP GROUP CALL PICK-UP}** button*

[PGM] + **{FLEX}** + '588' + **[SAVE]**

To Pick-up a call ringing at another station

1. Lift the handset or press **[SPEAKER]**.
2. Dial '588', the Pick Up Group Call Pick-up feature code.

Or,

2. Press the **{PICK UP GROUP CALL PICK-UP}** button.

SLT

To Pick-up a call ringing at another station

1. Lift the handset
2. Dial '588', the Pick Up Group Call Pick-up code.
3. Dial Pick Up Group Number.

Conditions

1. When a user dial 'Group Call Pick-up feature code(**)', a station can answer calls ringing at another station in the same pick up group in PGM190 and PGM192. But when a user dial 'Pick Up Group Call Pick-up feature code(588)', a station can answer calls ringing at another station in

the same pick up group in PGM192. A station can't pick up the call of pick up group in PGM190 with 'Pick Up Group Call Pick-up feature code(588)'.

2. To pick-up a CO/IP call, the station must have an idle appearance button available.
3. When several calls are ringing simultaneously, Call Pick-up will connect the oldest highest priority call (Ringing Line Preference, PGM 173).
4. Queue callback and Private Line calls are not subject to Call Pick-up and such attempts receive error tone.
5. Handsfree announced intercom calls cannot be picked up by another station. Only ringing intercom calls are subject to Call Pick-up.
6. When a station belongs to multiple pick up groups, calls to the group with the lowest number are answered first.
7. It can be picked up by pressing flex button of associate station, if it's programmed ON at ADM 114-B20.

Programming

STATION GROUPS	1. Pick Up Group (PGM 192-Btn 1)
STATION	1. Pick up by Button(PGM 114-Btn 20)
SYSTEM	1. Ringing Line Preference Priority (PGM 173)

Related Features

Intercom Signaling Mode
Ringing Line Preference
Directed Call Pick-Up
Pick Up Group
Private Line

Hardware

2.16 AUTO CALL RECORDING

Description

iPECS Phone users can be configured in the system to record all calls to a mailbox or the hard disk drive of an iPECS Phontage or UCS Client. When recorded to a mailbox, users manage the recording through voice mail. For recordings to the Phontage or UCS Client, recordings are managed directly by Phontage or UCS Client with the ability to listen to, delete or send the recording to others via e-mail.

Operation

Recording of calls is automatic when assigned.

To manage the recordings, use the procedures outlined in the Phontage or UCS Client User Guide.

Conditions

1. The Phontage or UCS Client can record one call at a time and must be idle. While recording, if the Phontage or UCS Client places or receives a call, recording terminates.
2. When call recording begins, the station will receive a Call Recording confirmation tone.
3. A remote Phontage or UCS Client will not support call recording.

Programming

STATION

1. Two-way Record Privilege (PGM 112-Btn11)
2. Auto Call Recording (PGM 112-Btn 20)
3. Call Recording Destination (PGM 112-Btn 21)
4. (Optional) without MCIM set 3-Way Conference Preference to LOCAL. (PGM 160-161 item 53)

Related Features

Hardware

iPECS Phone, PC with Phontage or UCS Client

2.17 CALL TRANSFER

2.17.1 Call Transfer, Station

Description

CO/IP calls can be transferred to other stations in the iPECS system. Calls can be transferred announcing the call (screened) or without an announcement (unscreened).

When a call is transferred, the Transfer Recall Timer is initiated. If the timer expires before the call is answered, the Hold Recall process is initiated.

Operation

iPECS Phone

While on a CO/IP call, Screened Call Transfer

1. Press **[TRANS]**.
2. Dial the station to receive the transfer.
3. At answer or splash tone, announce the call.
4. Hang-up to complete the transfer.

Or,

1. Press the **{DSS/BLF}** button for the desired station.
2. At answer or splash tone, announce the call.
3. Hang-up to complete the transfer.

While on a CO/IP call, Unscreened Call Transfer

1. Press **[TRANS]**.
2. Dial the station to receive the transfer.
3. Hang-up to complete the transfer.

Or,

1. Press the **{DSS/BLF}** button for the desired station.
2. Hang-up to complete the transfer.

SLT

While on a CO/IP call, Screened Call Transfer

1. Momentarily depress the hook-switch.
2. Dial the station to receive the transfer.
3. At answer or splash tone, announce the call.
4. Hang-up to complete the transfer.

While on a CO/IP call, Unscreened Call Transfer

1. Momentarily depress the hook-switch.
2. Dial the station to receive the transfer.
3. Hang-up to complete the transfer.

Conditions

1. The transferring station may camp a call on to a busy station.
2. The LED of a **{LOOP}** or **{POOL}** button will display the status of a call until the station no longer has call supervision i.e.; the call is successfully transferred.
3. To prevent Toll abuse, CO/IP lines without an active call (either incoming or dialed digits on outgoing) cannot be transferred.
4. For outgoing CO Line calls, the system will monitor the CO Line for dial tone to prevent Toll abuse. When an IP Line is seized, the system does not monitor for dial tone.

Programming

STATION	1. No Touch Answer (PGM 111-Btn 6)
SYSTEM	1. Transfer Recall Timer (PGM 180-Btn 7)

Related Features

Hold Recall
Call Transfer, CO/IP
Call Waiting/Camp-On
Station Flexible Buttons

Hardware

2.17.2 Call Transfer, CO/IP

Description

A station may be permitted to transfer a CO/IP call to another CO/IP line, establishing an Unsupervised Conference between the two external parties.

If the receiving party is called through an ISDN or VoIP path, the Transfer Hold Recall Timer is initiated and if it expires, Hold Recall is initiated.

Operation

iPECS Phone

While on a CO/IP call, Screened Call Transfer

1. Press **[TRANS]**.
2. Place CO/IP call in normal manner.
3. At answer, announce the call.
4. Hang-up to complete the transfer.

While on a CO/IP call, Unscreened Call Transfer

1. Press **[TRANS]**.
2. Place CO/IP call in normal manner.
3. Hang-up to complete the transfer.

SLT

While on a CO/IP call, Screened Call Transfer

1. Momentarily depress the hook-switch.
2. Place CO/IP call in normal manner.
3. At answer, announce the call.
4. Hang-up to complete the transfer.

While on a CO/IP call, Unscreened Call Transfer

1. Momentarily depress the hook-switch.
2. Place CO/IP call in normal manner.
3. Hang-up to complete the transfer.

Conditions

1. For this feature, at least one of the two CO/IP lines (transferred or receiving) must provide detection of disconnect supervision and lost loop condition.
2. ISDN and VoIP calls can provide 'Answer Supervision', thus the iPECS system can provide Transfer Recall.
3. If, during the transfer to an external party, the user presses the CO/IP line of the original call, the outgoing call is disconnected and the original call is connected to the user.

4. An H.323 or SIP call cannot be transferred to a like VoIP call using unscreened call transfer.

Programming

STATION 1. Off Net Forward (PGM 111-Btn 14)

SYSTEM
1. Transfer Recall Timer (PGM 180-Btn 7)
2. Off Net Prompt Usage (PGM 160-Btn 11)
3. CO to CO US Timer Extension (PGM 160-Btn 12)
4. Unsupervised Conference Timer (PGM 182-Btn 5)

Related Features

Hold Recall
Call Transfer, Station
Unsupervised Conference

Hardware

2.17.3 Call Transfer, Voice Mail

Description

CO/IP calls can be transferred directly to a station's voice mailbox, either VMIM/VSF or an external Voice Mail system.

Operation

iPECS Phone

While on a CO/IP call, Call Transfer

1. Press **[TRANS]**.
2. Press **[MSG/CALLBK]** button.
3. Dial the number or press the **{DSS/BLF}** button for the desired station.
4. Hang-up to complete the transfer.

Conditions

1. The LED of a **{LOOP}** or **{POOL}** button will display the status of a call until the station no longer has call supervision i.e.; the call is successfully transferred.

Programming

STATION 1. Direct Transfer Mail Box Destination (PGM 120-Btn 6)

SYSTEM 1. Transfer Recall Timer (PGM 180-Btn 7)

Related Features

Hold Recall
Call Waiting/Camp-On
External Auto Attendant/Voice Mail
VMIM/VSF Voice Mail

Hardware

iPECS Phone

2.18 CALL WAITING/CAMP-ON

Description

Call Waiting is used to notify a busy station that a call is waiting to be answered. The busy station is notified of the waiting call by a 'Camp-On' tone. For users of an iPECS Phone, the LED of the **[HOLD]** button will flash.

After receiving a busy signal, the calling station camps on to the called station. The called station can respond by:

a. answering the waiting call, which places the active call on hold first,
b. sending a silent text or voice message,
c. activating One-Time DND, or
d. ignoring the Camp-On tone.

Operation

iPECS Phone

To activate a Camp-On while receiving Intercom busy tone

1. Press the '*' button, called and calling stations receive Camp-On tone.

SLT

To activate a Camp-On while receiving Intercom busy tone

1. Press the '*' button, called and calling stations receive Camp-On tone.

Conditions

1. The user may only Camp-On to a station in the busy mode. A user may not Camp-On to a station in DND, in a conference, receiving a Page, etc.
2. The Camp-On procedure is employed by an Attendant or Secretary of an Executive/Secretary pair to activate DND Override.
3. A Camp-On tone is sent each time the calling user presses the '*' button.

4. A text or voice message, Pre-defined or Custom may be used to respond to a Camp-On.

Programming

Related Features

Pre-defined & Custom Text Display Messages
DND (Do Not Disturb)
Intercom Call (ICM Call)
Silent Text Message
Voice Over

Hardware

2.19 CO/IP ACCESS

Description

Stations can access outgoing CO/IP lines based on CO/IP Group Access programming. iPECS Phones may use flexible buttons assigned to access a specific **{CO}** line, **{POOL}** button for outgoing calls or a **{LOOP}** button.

Individual users may be allowed to assign CO/IP access flexible buttons.

Operation

iPECS Phone

*To assign a **{LOOP}** button*

[PGM] + **{FLEX}** + **[PGM]** + '85' + **[SAVE]**

*To assign a **{POOL}** button*

[PGM] + **{FLEX}** + '8x' + **[SAVE]**

To place an outgoing CO/IP call

1. Lift the handset or press the **[SPEAKER]** button.
2. Press desired **{CO}** line, **{POOL}**, **{LOOP}** button or dial the CO line or Group access code.

To answer an incoming CO/IP call

1. Lift the handset or press the **[SPEAKER]** button.

Or,

1. Press flashing **{CO}** line, **{POOL}** or **{LOOP}** button; lift the handset to speak privately

SLT

To place an outgoing CO/IP call

1. Lift handset.
2. Dial the CO line or Group access code.
3. Dial the desired number.

To answer an incoming CO/IP call

1. Lift handset.

Conditions

1. When a user dials '9', the system will search the preferred CO group for idle CO/IP line, if there is no idle CO/IP line then the system will search the 1st CO/IP Group for an idle CO/IP line. The system may continue the search through all CO/IP line groups for an available line if "1st CO/IP Group Override" is enabled.
2. A telephone user not allowed access to a CO/IP line receives error tone when access is attempted. The station may receive transferred calls on such denied access lines but will not be able to flash or use the CO/IP line for an outgoing call.
3. A station denied access to a CO/IP line but assigned to have ring for the CO/IP line will receive ring, a flashing **{CO}**/**{IP}** line LED and may answer the call. The user may transfer the call but cannot make an outgoing call on the CO/IP line.
4. CO/IP lines placed on hold may be retrieved by dialing the "retrieve held CO/IP code '8#' and the CO/IP line number.
5. The Tx path to a station will be muted until the system has verified the Toll Restriction for the CO/IP line.
6. When a CO line is seized, the system will monitor the line for dial tone.
7. The system selects lines from a group using the Round robin or Last-choice method based on Admin Programming.
8. When an iPECS Phone is not assigned Ringing Line Preference, the user must press the ringing **{CO}**/**{IP}** line button to answer the call.
9. iPECS Phone users may be assigned exclusive use of a CO line as a Private Line.

Programming

STATION	1. CO/IP Line Access (PGM 112-Btn 4)
	2. CO Line Programming (PGM112-Btn 6)
	3. Prefer CO/GROUP (PGM112-Btn 14th)
	4. CO Line Group Access (PGM 117)
CO/IP	1. CO/IP Line Group (PGM 141-Btn 1)
	2. CO/IP Ring Assignment (PGM 144)

Related Features

CO/IP Line Groups
CO/IP Ring Assignment
Ringing Line Preference
Private Line

Hardware

2.20 CO/IP CALL TIME RESTRICTION

Description

The system can be programmed to limit the length of outgoing calls by specified stations. When a specified station places an outgoing call, the system initiates the Call Restrict timer and, 15 seconds prior to expiration, a warning tone is delivered. At expiration, the system terminates the call returning the external CO/IP line to idle.

In PGM 142 – 21 : CO CUT OFF TIMER is added.
By using this timer, CO Base call time restriction is possible.

Operation

System

Operation of this feature is automatic when assigned

Conditions

1. Once activated, the Call Warning Tone timer continues timing while the call is connected to the system even if the call is transferred or picked up at another station.

2. There are two kinds of CUT OFF timer. One is Station base, the other is CO base(It is newly added). If two timers are assigned simultaneously, call can be followed by shorter timer.

Programming

STATION	1. CO/IP Call Time Restriction (PGM 112-Btn 3) 2. Cut Off Timer (PGM 123-Btn 2)
CO/IP	1. CO CUT OFF TIMER (PGM 142-Btn 21)

Related Features

CO/IP Call Warning Tone Timer

Hardware

2.21 CO/IP CALL WARNING TONE TIMER

Description

Stations can receive a tone indicating the elapsed time of a CO/IP call has reached the Warning Tone time. The station hears the tone after the CO Warning Tone Timer has elapsed.

Operation
System

Operation of this feature is automatic when assigned

Conditions
1. Warning tone is received 15 seconds prior to expiration of the timer and is repeated at intervals equal to the Warning Tone Timer.

Programming
STATION	1. CO Call Time Tone (PGM 112-Btn 1)
SYSTEM	1. Elapsed Call Tone Timer (PGM 180-Btn 19)

Related Features
CO/IP Call Time Restriction

Hardware

2.22 CO/IP QUEUING

Description
When CO/IP lines are busy, permitted users can request to be placed in queue awaiting the CO/IP line or a CO/IP line in the same group to become available. When an appropriate CO/IP line becomes available, the system calls the waiting station on a first in first out basis.

Operation
iPECS Phone

To request to be placed in queue for a busy CO/IP line
1. Press busy **{CO/IP}** or **{CO/IP GRP}** button.
2. Press the **[MSG/CALLBK]** button, confirmation tone is received.
3. Hang-up, the **[MSG/CALLBK]** LED flashes.

To cancel the queue from the queued station
1. Press the **[MSG/CALLBK]** button, the **[MSG/CALLBK]** LED extinguishes.

SLT

To request to be placed in queue while receiving "All Lines Busy" signal

1. Momentarily press the hook-switch.
2. Dial '556', the Callback feature code.

To cancel the queue from the queued station

1. Lift the handset.
2. Dial '556', the Callback feature code.

System

When a CO/IP line becomes available

1. Send distinctive Queue recall to the station with oldest queue, flash appropriate **{CO}**/**{IP}** line button LED. Make CO/IP line and station busy to all other users.

Conditions

1. A CO/IP line can have any number of simultaneous queue requests.
2. A station may only have a single active CO/IP queue request. Activating a new queue request will replace, and thus cancel, an existing queue.
3. A Queue recall will always signal the station with tone ring, ignoring the station's assigned Intercom Signaling mode.
4. Queue recall will bypass a busy station, and place the station at the bottom of the queue list.
5. Queue recall will signal a station for 15 seconds, after which, the station is removed from the queue; the queue is cancelled.

Programming

STATION 1. CO/IP Queuing (PGM 112-Btn 5)

Related Features

CO/IP Access

Hardware

2.23 CONFERENCE

2.23.1 Conference Room

Description

In addition to ad-hoc conferencing, users may establish a Conference Room. Other internal and external parties are invited to the conference and can join the conference without further action by the user who established the Conference Room. A user can transfer an active call to a Conference

Room. A Conference Room can be password protected so that only parties that enter the password are allowed to join the Room.

Up to 9 Conference Rooms can be set-up and each can support a maximum of 32 parties with the g.711 or g.729 codec or 24 parties with the g.723 codec. Conference Rooms employ channels from an MCIM (Multi-party Conference Interface Module). Each MCIM supports up to 32 parties and multiple MCIM units may be installed as shown in the chart below.

iPECS-Micro	1 MCIM unit maximum
iPECS-50 & 100	2 MCIM units maximum
iPECS-300	4 MCIM units maximum
iPECS-600	8 MCIM units maximum
iPECS-1200	30 MCIM units maximum

Operation

Attendant Phone

To view Room participant list

1. Press the **PGM** button.
2. Dial '054'.
3. Dial Room number (1~9).

To delete a Conference Room

1. Press the **PGM** button.
2. Dial '055'.
3. Dial Room number (1~9).

iPECS Phone

To set-up a Conference Room

1. Press the **PGM** button.
2. Dial 53 to create a Conference Room.
3. Dial the desired Conference Room number (1~9).
4. If desired, enter a password for the Conference Room (must be exactly 5 digits).
5. Press **[SAVE]** to establish the Room.

To join a Conference Room

1. Dial *59, the Conference Room entry code.
2. Dial the Conference Room Number.
3. Dial the Conference Room password.

To delete a Conference Room

1. Press the **[PGM]** button.
2. Dial 54, the delete Conference Room code.
3. Dial the Conference Room number (1~9)
4. Dial the Conference Room password.
5. Press **[SAVE]** to delete the Conference Room.

To transfer a call to a Conference Room

1. Press the **[TRANS]** button.
2. Dial 59, the Conference Room entry code.
3. Dial the Conference Room Number.
4. Dial the Conference Room password
5. Hang-up to complete the transfer.

SLT

To set-up a Conference Room

1. Lift the handset
2. Dial 561, SLT Programming code.
3. Dial 53, the Create Conference Room code.
4. Dial the desired Conference Room number (1~9).
5. Dial the Conference Room password.
6. Momentarily press the hook-switch.

To join a Conference Room

1. Lift the handset.
2. Dial *59, the Conference Room entry code.
3. Dial the Conference Room Number.
4. Dial the Conference Room password.

To delete a Conference Room

1. Lift the handset.
2. Dial 561, SLT Programming code.
3. Dial 54, the delete Conference Room code.
4. Dial the Conference Room number (1~9).
5. Dial the Conference Room password.
6. Momentarily press the hook-switch, receive confirmation tone.

Conditions

1. Once established, a Conference Room will remain opened until the Room is deleted.
2. The Conference Room feature employs the Multi-party Conference Interface Module (MCIM), which must be installed for proper operation.
3. iPECS Phontage and UCS Client may also create, delete and join a Conference Room. For operation, refer to the iPECS Phontage or UCS Client User Guide.

Programming

Related Features

Multi-Party Voice Conference
Automatic Speaker Select
Hold Recall
Unsupervised Conference

Hardware

MCIM, Multi-party Conference gateway Module

2.23.2 Multi-Party Voice Conference

Description

The system allows multiple internal and external parties to be connected on a call, conference. An unlimited number of 3-party conferences may be established using iPECS Phones. In addition, each MCIM (Multi-party Conference Interface Module) up to 32 parties with the g.711 or g.729 codec and 24 with the g.723 codec may be connected in a single voice conference. The MCIM will support any combination of parties and conferences to the maximum total number of parties in conference. Multiple MCIMs, see chart below, are installed to support multiple multi-party conferences with a maximum of 32 parties in any single conference.

iPECS-Micro	1 MCIM unit maximum
iPECS-50 & 100	2 MCIM units maximum
iPECS-300	4 MCIM units maximum
iPECS-600	8 MCIM units maximum
iPECS-1200	30 MCIM units maximum

Operation

iPECS Phone

To establish an ad-hoc conference

1. Establish first call.
2. Press the **[CONF]** button. The LED will light, the connected party is placed on exclusive hold and the user receives dial tone.
3. Place second call.
4. When connected, press **[CONF]**, new call is placed on exclusive hold.
5. Repeat steps 3 and 4 above to add additional conference parties.
6. Press **[CONF]** button to establish conference.

To place a conference on hold

1. Press the **[HOLD]** button, the **[CONF]** button LED will flash.

To retrieve held conference

1. Press **[CONF]** button, all parties reconnected.

SLT

To establish an ad-hoc a conference

1. Establish first call.
2. Momentarily press the hook-switch, the connected party is placed on exclusive hold and the user receives dial tone.
3. Place second call.
4. When connected, repeat steps 2 and 3 above to add additional parties.
5. When completed adding parties, momentarily press hook-switch twice in 2 seconds, all parties are connected.

Conditions

1. The **[CONF]** button remains illuminated at the initiators phone for the duration of the conference.
2. There is no limit on the number of 3-way conferences the system will support with iPECS Phones. With the MCIM, the system can support conferences of more than 3 parties. Each MCIM supports 32 conference channels with the g.711 or g.729 codec or 24 with the g.723 codec and can support multiple conferences with a total maximum number of 32 or 24 users.
3. If all MCIM channels are busy when a request for a channel is received, the user will receive error tone and the display will indicate that no Conference channels are available.
4. If the system receives a disconnect signal and no internal parties remain in the conference, the conference is terminated and all parties are disconnected. If an internal party is still connected when a disconnect signal is received, the connection to remaining parties is maintained.
5. The normal Hold Recall process is applied to a conference on hold using the Unsupervised Conference recall Timer for recall timing.
6. If while setting up a conference, system error tone is received, the initiator must press the **[CONF]** button (SLT must hook-flash) to regain Intercom dial tone.
7. A station that is busy, in DND or other non-idle state cannot be added to a conference.
8. The MCIM is also employed to support the Conference Room feature.

Programming

Related Features

Automatic Speaker Select
Hold Recall
Unsupervised Conference
Broker Call
Conference Room

Hardware

MCIM required to support voice conferences with more than 3 parties.

2.23.3 Unsupervised Conference

Description

An iPECS phone user may establish a conference with external parties and exit the conference while allowing the external parties to converse privately without supervision from the user.

The system will disconnect the Unsupervised conference if disconnect is detected with only two parties connected or at expiration of the Unsupervised Conference timer. Fifteen seconds prior to expiration of the timer, a Disconnect Warning Tone is provided.

If enabled, either party in an Unsupervised Conference can request the Unsupervised Conference timer be extended. The party enters the Timer Extension feature code and a digit 1 to 9 indicating the Timer extension multiplier. The system will then extend the timer based on the dialed digit multiple of the Timer. For example, if the Unsupervised Conference timer is 5 minutes and the user dials the digit 4, the timer will extend to 20 minutes (4 multiplied by 5 minutes).

Operation

iPECS Phone

To set up an Unsupervised conference

1. Establish normal conference.
2. Press the **[CONF]** button. The **[CONF]** button LED flashes to indicate the Unsupervised Conference state.

To reenter the Unsupervised conference

1. Press the flashing **[CONF]** button.

Conf Party

To extend the Unsupervised Conference from a connected party

1. Dial '* # #'.
2. Dial the Timer extension multiplier (1~9).

Conditions

1. The Unsupervised Conference Timer also applies to an external call placed by a DISA user.
2. An Unsupervised conference will be terminated if the system receives a disconnect signal or the Unsupervised Conference Timer expires.
3. An Unsupervised Conference will not recall the user.

Programming

NUMBERING	1. Unsupervised Conference Timer Extension Code (PGM 109-Btn 3)
STATION	1. Off Net Forward (PGM 111-Btn 14)
SYSTEM	1. CO to CO UC Time Extension (PGM 160-Btn 12)
	2. Unsupervised Conference Timer (PGM 182-Btn 5)

Related Features

Direct Inward System Access (DISA)
Multi-Party Voice Conference

Hardware

iPECS Phone to establish Unsupervised Conference
MCIM required to support conferences with more than 3 parties

2.23.4 Conference Group

Description

Stations and external contacts can be arranged in groups so that a user may create a conference with all members of the group through a single call. Each conference group can have up to 32 members that can be a station or an external telephone number. Conference groups are assigned in the iPECS User Portal (Station Web Admin) by assigning a conference group number and the members of the group. A voice recording, which is played to members receiving the Conference group call, can be associated with the group. Initiation of a Group call can be password protected with a 5-digit password assigned when the group is created.

There are two (2) types of Conference Groups, Station and System. The Station Conference group is created by the station user, who is assigned as the group supervisor, and is the only member that can initiate a Conference group call. The System Conference Group is created by an Attendant or system administrator and any member can initiate the Conference Group call. The member initiating the call becomes the supervisor for that Conference Group call.

The supervisor can manage membership in the conference and can monitor the status of each member, in/out of conference. The Supervisor can remove members from the conference or, if members are absent, the supervisor can add a non-member to the conference. An Absent Supervisor timer terminates the conference if the supervisor is not in the conference for the timer interval. Setting the interval to ‘0’ disables this feature.

Once a group is created, the supervisor imitates the conference by calling the conference group. The system then attempts to call each member of the group either simultaneously or in turn using the Interval timer assigned for the group. Members that answer the call receive the recorded voice announcement, if any, and can enter the conference, dial ‘1’, or reject the conference, hang-up. The system will attempt to contact a busy or no-answer member based on the assigned Retry Count. A call is consider no-answer using the no-answer timer set when the group was created.

Operation

<u>To create a conference group from the iPECS User Portal:</u>

1. Access Station Program in the Web Admin.
2. Select Conference Group button.

3. Create the Conference group entering:
   - The name of group : up to 12 character,
   - The password: Passwords must be 5-digits.
   - Announcement number for the group: (pre-recording the announcement is required),
   - The Absent Supervisor timer,
   - The No answer timer,
   - The busy/no-answer Retry count,
   - The Interval time: Each member is called in turn at expiration of the interval timer.
4. Set the Member Attributes by entering:
   - Index number
   - Type: Station, Individual CO, CO group, loop or Transit-out code (access loop for networking call).
   - The CO value: the supervisor wants to select a specific CO line or CO group.
   - Dial number: The station number or Co dial number.

IPECS phone

*To assign a **{GRP MONITOR}** button*

[PGM] + **{FLEX}** + **[PGM]** + 55' + **[SAVE]**

To initiate a Conference Group call:

1. Lift the handset
2. Dial ***68xxx, the Group Conference code and group number.
3. Enter the Conference group password, if assigned, members receive the Conference Group call.

To enter a Conference Group call:

1. Answer the ringing call and receive recorded announcement, if assigned and recorded.
2. Dial '1' to enter the conference or hang-up to reject the conference. For iPECS Phontage or UCS clients, use Accept or Deny when notified of the conference.

To monitor member conference status or delete members while in the conference:

1. Press the Monitor flex button, the state of the first member is displayed.
2. Press **[VOL UP/VOL DWN]** button to display the state of other members.
3. Press the **[DND]** button to delete a participant from the conference.

To monitor the Group member conference status while Not in the conference:

1. Press the **[PGM]** button
2. Dial 55, the Monitor Conference Group code.
3. Press **{VOL UP/VOL DWN}** button to display the state of other members.

To add a non-member to the Group:

1. Press the **[HOLD]** button to place the conference on hold.
2. Call the desired party.
3. Press the **[CONF]** button.

Conditions

1. The conditions associated with Multi-Party Voice Conference apply.
2. Members using the iPECS Phontage or UCS client may view the state of other members in the conference.
3. The number of available groups is given in Table 1.2-1 System Capacity Chart. Each group can have up to 32 members including the supervisor.
4. The user must have access to the Station Web Admin with the proper password (Authentication code).
5. If there are insufficient MCIM channels available when initiating the conference group call, the initiator receives error tone.
6. Two-way record can be used to record the Conference Group call.

Programming

Related features

Multi-Party Voice Conference
Conference Room
System Admin Programming

Hardware

MCIM

2.23.5 Conference Member VIEW & DELETE in Add_On_Conference

Description

In Add_On_Conference,
Conference supervisor can see the conference member and remove the member from conference.

Operation

1. Conference supervisor press {VIEW} soft button in conference.
2. LCD displays the first member. If member is a station, “STA XXX” display. If member is a co line, telephone number of caller/called party in the co line display.
3. Conference supervisor can see other members by pressing [Volume up/down] button. (Next member if [Volume down] button, previous member if [Volume up] button)
4. If press {DELETE} soft button, the current member will be removed from conference.
5. If press {BACK} soft button, conference supervisor will be back to normal conference state.

LCD of supervisor in conference.

CONFERENCE		
09 SEP 10		03:32 am
CONF	**VIEW**	MUTE

Press {VIEW} soft button.

MEMBER 01	
STA 101	
BACK	**DELETE**

Press [Volume up/down] to see next member.

MEMBER 02	
01223456789	
BACK	**DELETE**

Press {DELETE} soft button to remove the current member from conference.

MEMBER 02	
STA 104	
BACK	**DELETE**

Press {BACK} soft button to go back to normal conference state.

CONFERENCE		
09 SEP 10		03:32 am
CONF	**VIEW**	MUTE

Conditions

1. Only supervisor of Add_On_Conference can use this feature.
2. Only keyset with 3 soft button can use this feature.

Programming

Related features

Multi-Party Voice Conference

Hardware

2.24 CUSTOMER SITE NAME

Description
A Customer Name, up to 23 characters, may be entered into the system database. The name is displayed on the SMDR and database outputs as well as during an Admin session.

Operation
System

Operation of this feature is automatic when a name is assigned

Conditions

Programming
SYSTEM ID 1. Customer Site Name (PGM 100-Btn 2)

Related Features

Hardware

2.25 DATA LINE SECURITY

Description
Data transmitted over analog CO lines is subject to distortion and errors if system tones such as Camp-On and Override are applied during transmission. To eliminate such errors, stations that use analog data (modems or Fax) can be assigned to block incoming system tones.

Operation
System

System tones are automatically blocked when Data Line Security is assigned

Conditions
1. Stations or an Attendant attempting to Camp-On or Override a station with Data Line Security will receive error tone.
2. When Data Line Security is enabled, the system will not apply audio gain to the call.

Programming
STATION 1. Data Line Security (PGM 111-Btn 4)

Related Features

Call Waiting/Camp-On
DND Override
Intrusion

Hardware

2.26 Delayed CO/IP Ring

Description

Ring signals for an incoming CO/IP call can be sent to stations immediately upon detection or after an assigned ring cycle delay. The delay can be up to 9 system ring cycles, thus allowing other stations to answer the call.

Operation

System

Delay Ring operation is automatic when assigned

Conditions

1. Delay Ring can be assigned for a station or Station Group.
2. If no delay is entered when programming Ring assignments, the station will receive immediate ring.
3. Private Lines may be assigned with delayed ring.
4. If no station or Station Group is assigned for immediate ring, the call will ring immediately at the first available Attendant.

Programming

CO/IP 1. CO Station Ring Assignment (PGM 144)

Related Features

Private Line
Station Groups
CO/IP Ring Assignment

Hardware

2.27 Delayed Auto Attendant

2.27.1 Ring Assigned Case

Description

An incoming CO/IP call can be routed to the VMIM/VSF Auto Attendant either immediately upon detection or after a delay of up to 30 seconds. This allows other stations assigned immediate ring the opportunity to answer before the call is routed to the Auto Attendant.

Operation

System

Operation of this feature is automatic when assigned

Conditions

5. When Delayed Auto Attendant Ring is assigned, after the delay, the call will no longer ring assigned stations and will only ring to the VMIM/VSF Auto Attendant.
6. If no delay is entered, the call will ring to the VMIM/VSF Auto Attendant immediately.
7. To assign Delayed Attendant ring, at least one station or Station Group must be assigned immediate ring.
8. Ring is assigned to a VMIM/VSF Auto Attendant announcement (01-70) as a "station type" with a delay from 00 to 30 seconds.

Programming

CO/IP 1. CO Station Ring Assignment (PGM 144)

Related Features

CO/IP Ring Assignment

Hardware

2.27.2 DISA Incoming Case (Russia Only)

Description

In case of DISA incoming call, there are two Timers that is related with DISA Delay.
These timers can be set in PGM 142 – Co-line Base.

DISA Answer Timer : When DISA incoming call is routed to a system, after this timer expired, system is answered.
DISA Delay Timer : When system is answered, after this timer expired, VSF system announcement start to play.

Operation

Conditions

1. These two timer is applied only for Russia.

Programming

CO/IP 1. CO Line Attribute 2(PGM 142)

Related Features

1. DISA Service Attribute
2. DID Service Attribute.

Hardware

2.28 DIAGNOSTIC/MAINTENANCE

Description

The system software incorporates various diagnostic and maintenance routines that may be “called” remotely or locally through the systems RS-232 serial ports, a TCP/IP connection using a Web browser established over IP networks or a PPP connection over ISDN. Routines that can be accessed include trace functions at the device level, commands for diagnostics and maintenance, and tools for manipulation at the OS level.

An optional Network Management System application is available providing remote access to the system for maintenance and diagnostics. For details on the iPECS NMS application, refer to the iPECS NMS Manual.

Operation

Conditions

Programming

Related Features

Hardware

2.29 DIAL-BY-NAME

Description
A name, up to 16 characters, may be assigned to each Station and System Speed dial. In addition, each station may be assigned a 12-character name. When assigned, a user may place an intercom call to another station or select a Station or System Speed dial using the name.

The user selects from one of three Dial-by-Name directories and enters characters employing 2 dial pad buttons for each character. The system finds and displays the nearest match to the user entries. The user may continue entering characters or scroll the directory at any point using the **[VOL UP]/[VOL DWN]** button and select a name to call. The number associated with a selected name is displayed by using the **[TRANS]** button.

Operation
iPECS Phone

*To use Dial by Name using a **[SPEED]** button*
1. Press the **[SPEED]** button twice
2. Dial the desired directory, I: Intercom, U: User Speed Dial, S: System Speed Dial.
3. Search the directory using the **[VOL UP]/[VOL DOWN]** button or by entering characters. See Station Speed Dial for character entry procedure.
4. Press the **[SAVE]** button to place the call.

To use Dial by Name using the LIP-8000 series soft buttons
1. Press the **[DIR]** soft button
2. Dial the desired directory, 1: User Speed Dial, 2: System Speed Dial 3: Intercom.
3. Search the directory using the **Navigation** button or by entering two digits for each character, see section 2.65.3 for character entry procedure.
4. Press the **[HOLD/SAVE]** button to place the call.

To toggle between the name and number display
1. Press the **[TRANS]** button.

To program the station user name
1. Press the **[PGM]** button.
2. Dial '74', the User Name Program code.
3. Dial the name, up to 12 characters; refer to Station Speed Dial for character entry.
4. Press **[SAVE]**.

SLT

To program the station user name

1. Lift the handset
2. Dial ‘561’, SLT Programming code.
3. Dial ‘74’, the User Name Program code.
4. Dial the name, up to 12 characters; refer to Station Speed Dial for character entry.
5. Momentarily press the hook-switch, confirmation tone is received.

System Attendant

To program a name for another station

1. Press the **[PGM]** button.
2. Dial ‘071’, the Attendant User Name Program code.
3. Dial the station number.
4. Dial the name, up to 12 characters; refer to Station Speed Dial for character entry.
5. Press **[SAVE]**.

Conditions

1. Available characters are A to Z, space and period.
2. The LCD will display multiple names, one per LCD line up to 16 characters.
3. If a user selects a directory with no entries or there is no match to the user entry, the “No Entries” message is displayed and error tone is provided.
4. Dial-by-Name is only available to iPECS Phones with a display. Other users will receive error tone if an attempt is made to access Dial-by-Name.
5. A user may both scroll and enter characters to search a directory.

Programming

STATION 1. System Speed Dial Access (PGM 106-Btn 16)

Related Features

Station Speed Dial
System Speed Dial

Hardware

iPECS Phone w/Display

2.30 DIAL PULSE TO TONE SWITCHOVER

Description

On a pulse dial CO line, the user can request the system to change the signaling mode from pulse to DTMF. This allows the user to access outside facilities that require DTMF signals such as banking services, voice mail, etc.

Operation

To switch from pulse to DTMF while on a pulse CO line

1. Dial '*', signaling changed to DTMF.

Conditions

1. In a Speed Dial, the '*' will automatically insert a pause before dialing the remaining digits.
2. This command is only recognized for analog pulse dial CO lines.
3. Dial pulse to tone switchover is not available in the Redial features.
4. When '*' is the first digit of a dialed number, the Display Security feature is enabled and not Pulse to tone switchover.
5. For VoIP calls, pulse dialing is not available thus; switchover is not required or supported.

Programming

CO/IP

1. CO Line Type (PGM 141-Btn 4)
2. CO Line Signal Type (PGM 141-Btn 5)

Related Features

Speed Dial
Display Security

Hardware

2.31 Dialing Restrictions

2.31.1 Class of Service

Description

Dialing privileges can be assigned for each station, DISA line and Authorization Code. The dialing privileges are the result of the interaction of the Station and CO Class of Service (COS) assignments as shown in the following tables. Users placing an outgoing call or dialing after answering a call will be allowed the dialing privileges assigned.

Station/DISA/Code Class of Service – There are eleven (11) Station Class of Service assignments, which also apply to DISA and Authorization Code users based on the assigned COS.

Station COS	Dialing Restriction
1	No restrictions are placed on dialing.
2	Assignments in Exception Table A are monitored for allow and deny numbers.
3	Assignments in Exception Table B are monitored for allow and deny numbers.
4	Assignments in Exception Tables A & B are monitored for allow and deny numbers.
5	The leading digit cannot be a Long Distance code and assignments in Exception Table C apply.
6	Number of digits cannot exceed LD digit count and assignments in Exception Table C apply.
7	Intercom and Emergency number calls are allowed. Incoming and transferred calls are allowed.
8	Assignments in Exception Table D are monitored for allow and deny numbers.
9	Assignments in Exception Table E are monitored for allow and deny numbers.
10	The assignments in the Exception Table D & E are monitored for allow and deny numbers.
11	The assignments in the Exception Table A & B and D & E are monitored for allow and deny numbers.

CO Class of Service – There are five (5) CO line COS assignments.

CO COS	Restriction
1	Dialing privileges are governed by the Station COS only.
2	Assignments in Exception Table A govern dialing privileges for Station COS 2 & 4.
3	Assignments in Exception Table B govern dialing privileges for Station COS 3 & 5.
4	No LD calls (LD code & digit count) and assignments of Exception C apply to Station COS 2 & 6 only.
5	No dialing restrictions applied, overrides Station COS 2 through 6.

Station & CO COS Interaction The following chart displays the dialing capabilities based on the interaction between the Station COS and CO line COS assignments.

Station COS	Restriction				
	CO COS 1	CO COS 2	CO COS 3	CO COS 4	CO COS 5
1	No restriction	No restriction	No restriction	No LD calls & Table C	No restriction
2	Exception Table A	Exception Table A	No restriction	No LD calls & Table C	No restriction
3	Exception Table B	No restriction	Exception Table B	No LD calls & Table C	No restriction
4	Exception Table A & B	Exception Table A	Exception Table B	No LD calls & Table C	No restriction
5	Local call & Table C	Local call & Table C	Local call & Table C	No LD calls & Table C	No restriction
6	Local call & Table C	Local call & Table C	Local call & Table C	No LD calls & Table C	No restriction
7	Intercom only	Intercom only	Intercom only	Intercom only	Intercom only
8	Exception Table D	Exception Table D	No restriction	No LD calls & Table C	No restriction
9	Exception Table E	Exception Table E	No restriction	No LD calls & Table C	No restriction
10	Exception Table D & E	Exception Table D & E	No Restriction	No LD calls & Table C	No Restriction
11	Exception Table A, B, D & E	Exception Table A, B, D & E	No Restriction	No LD calls & Table C	No Restriction

PBX Dialing Codes – Four (4) one or two-digit PBX Trunk Access Codes can be defined in the system database. When dialed as the first digit(s), these codes signal the system to apply the appropriate COS. If not dialed, the call is treated as an internal PBX call and dialing is not restricted.

Exception Tables – Each Exception Table permits entry of 50 Allow codes and 50 Deny codes. Each code can contain up to 20 digits including digits 0-9, "#" as a wild card (any digit) and "*" as the end of entry mark. Reference the previous charts for application of the Exception Tables.

Exception Table process – As digits are dialed, they are compared to entries in the appropriate Exception Table. Based on the Allow and Deny entries, the system applies the following rules to allow or deny the call.

Rule 1 – If a table has no entries, no restrictions are applied.
Rule 2 – If there are only Deny entries, restrictions are provided as Deny only.
Rule 3 – If there are only Allow entries, restrictions are provided as Allow only.
Rule 4 – If there are both Allow and Deny entries, the Deny entries are searched. If the dialed number matches a Deny entry, the call is restricted; if no match is found the call is allowed.

Operation

System

The system automatically applies the assigned COS

Conditions

1. The dialing privileges are based on the CO and Station COS.
2. The selection of the COS is made based on the Authorization code entered by the user.
3. COS does not apply to calls placed over a VoIP channel.

Programming

STATION	1. Station COS (PGM 116)
CO/IP	1. CO COS (PGM 141-Btn 2)
SYSTEM	1. DISA COS (PGM 166) 2. PBX Trunk Access Codes (PGM 172) 3. LD Digit Count (PGM 177-Btn 4 4. LD Digit Code (PGM 177-Btn 19)
TABLES	1. Toll Restriction (PGM 224)

Related Features

Day/Timed & Night Station COS

Direct Inward System Access (DISA)
Temporary Station COS/Lock
Walking COS

Hardware

2.31.2 Day/Timed & Night Station COS

Description

Each station, DISA line and Authorization code is assigned a COS for two modes: Day, which includes Timed, and Night service modes. The service mode is generally controlled by the System Attendant and, based on the mode, appropriate dialing privileges are established.

Operation

System

Dialing restrictions are automatically applied based on COS assignments

Conditions

Programming

STATION	1. Station COS (PGM 116)
CO/IP	1. CO COS (PGM 141-Btn 2)
SYSTEM	1. DISA COS (PGM 166) 2. PBX Trunk Access Codes (PGM 172) 3. LD Digit Count (PGM 177-Btn 4) 4. LD Digit Code (PGM 177-Btn 19)
TABLES	1. Toll Restriction (PGM 224) 2. Authorization Codes Table (PGM 227)

Related Features

Authorization Codes (Password)
Class of Service
Direct Inward System Access (DISA)
Temporary Station COS/Lock
Walking COS
Auto Service Mode Control
Day/Night/Timed/Scenario Ring Mode

Hardware

2.31.3 Temporary Station COS/Lock

Description

A user or an Attendant can change the Station's Class of Service to COS 7, temporarily preventing unauthorized toll dialing from the station, i.e. "lock the station". The station is still allowed to place internal calls and Emergency number calls.

Operation

iPECS Phone

To activate Temporary COS

1. Press the **[PGM]** button.
2. Dial '21', the Temp COS code.
3. Press the **[SAVE]** button.

To restore the assigned COS

1. Press the **[PGM]** button.
2. Dial '22', the restore COS code.
3. Dial Authorization Code (station number and code).
4. Press the **[SAVE]** button.

SLT

To activate Temporary COS

1. Momentarily press the hook-switch.
2. Dial '561', the SLT Programming code.
3. Dial '21', the Temp COS code.
4. Momentarily press the hook-switch.

To restore the assigned COS

1. Momentarily press the hook-switch.
2. Dial '561', the SLT Programming code.
3. Dial '22', the restore COS code.
4. Dial Authorization Code (station number and code).
5. Momentarily press the hook-switch.

System Attendant

To activate Temporary COS

1. Press the **[PGM]** button.

2. Dial '021', the Temp COS code.
3. Dial station range.
4. Press the **[SAVE]** button.

To restore the assigned COS

1. Press the **[PGM]** button.
2. Dial '022', the restore COS code.
3. Dial station range
4. Press the **[SAVE]** button.

Conditions

1. The station is restored to the Station COS as appropriate for the active service mode, Day, Night, or Timed.

Programming

STATION	1. Station COS (PGM 116)
CO/IP	1. CO COS (PGM 141-Btn 2)
SYSTEM	1. DISA COS (PGM 166)
	2. PBX Trunk Access Codes (PGM 172)
	3. LD Digit Count (PGM 177-Btn 4)
	4. LD Digit Code (PGM 177-Btn 19)
TABLES	1. Toll Restriction (PGM 224)
	2. Authorization Codes Table (PGM 227)

Related Features

Class of Service
Walking COS
Auto Service Mode Control
Day/Night/Timed/Scenario Ring Mode
Authorization Codes (Password)

Hardware

2.31.4 Walking COS

Description

A user may temporarily override the toll restriction of a station to make toll calls from a normally toll restricted station. The user must input an Authorization Code in order to activate Walking COS and

is subject to the assigned COS of the Authorization Code. The COS associated with the Authorization Code is applied after the code is entered for the next call.

Operation

iPECS Phone

To activate Walking COS

1. Press the **[PGM]** button.
2. Dial ‘23’, the Walking COS code.
3. Dial the station number and Authorization code, ‘*’ and System Authorization code or “*”, the code index and System Authorization code.
4. Place call as normal.

SLT

To activate Walking COS

1. Momentarily press the hook-switch.
2. Dial ‘561’, the SLT Programming code
3. Dial ‘23’, the Walking COS code.
4. Dial the station number and Authorization code, ‘*’ and System Authorization code or “*”, the code index and System Authorization code.
5. Place call as normal.

Conditions

1. The Station COS applied for Walking COS is the COS of the station associated with the Authorization code or, for a System Authorization code, COS1.
2. Walking COS applies the COS for only one call. Terminating the call returns the station to the assigned Station COS. The user may reactivate Walking COS to place another call or use Flash to maintain the Walking COS.
3. Use of Authorization codes varies based on the system nation code. In some regions, particularly the US and UK, a System Authorization code may be required for DISA access. Entering a Station code on a DISA line will fail in these areas. In other regions, a System Authorization code can be used with Walking COS.

Programming

STATION	1. Station COS (PGM 116)
CO/IP	1. CO COS (PGM 141-Btn 2)
SYSTEM	1. DISA COS (PGM 166) 2. PBX Trunk Access Codes (PGM 172) 4. LD Digit Count (PGM 177-Btn 4) 3. LD Digit Code (PGM 177-Btn 19)
TABLES	1. Toll Restriction (PGM 224) 2. Authorization Codes Table (PGM 227)

Related Features

Class of Service
Auto Service Mode Control
Day/Night/Timed/Scenario Ring Mode
Authorization Codes (Password)

Hardware

2.32 DIFFERENTIAL RING

Description

Differential Ring allows any one of 14 different audible Ring signals to be assigned to an iPECS Phone, allowing users to determine which phone is ringing and the type of call (Intercom or CO/IP). When the phone receives an incoming call, the selected ring signal is provided over the speaker. Different selections are assigned for Intercom and CO/IP calls.

Eight different tones are stored in the iPECS Phone. Four of these tones are permanent while the other four are assigned from the 10 ring-tones in the iPECS system. Note the system ring-tones may be replaced with any 8 second **.wav* file through the iPECS Web Maintenance.

Operation

iPECS Phone

To select the desired ring tone

4. Press the **[PGM]** button.
5. Dial '1' for Ring Selection.
6. Dial '1' for Intercom or '2' for CO/IP ring
7. Dial Ring Tone selection '1'-'8', ring tone is received.

To select the desired ring tone By Co-line Base.

1. Enter PGM 141 – F16.
2. Dial desired Ring tone.
3. 00 is Not used for Co-line base, 01-12 can be selected.

To download a Ring tone from the system to an iPECS Phone

1. Press the **[PGM]** button.
2. Dial 1 for Ring selection.
3. Dial 5 for Ring tone download.
4. Dial Ring tone storage bin '5'-'8'.
5. Dial Ring tone selection, '0'-'9', tone is received.
6. Press the **[SAVE]** button.

Conditions

1. To employ one of the system Ring tones, it must first be downloaded to a Ring tone storage bin in the iPECS Phone.
2. The iPECS Phontage and UCS Client do not have access to the system Ring tones. In the iPECS Phontage or UCS Client multiple tones are available and the user may load an *.wav file for use as a Ring tone.
3. Any or all of the ten ring-tones stored in the system can be replaced via the iPECS Web Maintenance.

Programming

Related Features

Hardware

iPECS Phone

2.33 DND (Do Not Disturb)

Description

A station, which is allowed Do Not Disturb, can be placed in DND to block incoming ring for CO/IP and Intercom calls, transfers and paging announcements.

Operation

iPECS Phone

To activate DND

1. Press the **[DND]** button, the **[DND]** button LED illuminates.

To remove DND

1. Press the **[DND]** button, the **[DND]** button LED extinguishes.

SLT

To activate DND

1. Dial '553', the DND feature code; stutter dial tone is received.

To remove DND

1. Dial '553', the DND feature code, dial tone is received.

Conditions

1. A station will receive error tone if not allowed access to DND.
2. If DND is allowed, pressing the **[DND]** button while ringing will activate One-Time DND.
3. Only the Secretary of and Executive/Secretary Pair or an Attendant may override DND at the Executive's station.
4. An Attendant may cancel DND for other stations.
5. DND service is not available to Attendants.
6. Recalls for CO/IP calls will override the DND feature.
7. A station in DND is out-of-service for all incoming calls including Station Group calls.
8. A station in DND is bypassed by calls forwarded to the station. If the last station in a Call Forward chain is in DND, the call will ring to the previous station in the chain.

9. When calling a station in DND, the iPECS Phone display will indicate the DND status.

Programming

STATION 1. DND (PGM 111-Btn 3)

Related Features

Feature Cancel
Intrusion
Call Forward
DND Override
Executive/Secretary Forward
Station Groups
DND - One Time DND

Hardware

2.34 DOOR OPEN

Description

The iPECS hardware is equipped with relays that activate External Control Contacts. The contacts can be assigned to one of several functions including a Door Open Contact. When used as a Door Open Contact, the contact is connected to a door-lock release mechanism. When assigned stations receive the Door Bell signal, the user may dial the Door Open code to activate the contact.

iPECS Phone users may assign a Flex button as a **{DOOR OPEN}** button.

Operation

iPECS Phone

To assign a {DOOR OPEN} button

[PGM] + **{FLEX}** + Door Open code ('#*1', '#*2', '#*3' or '#*4') + **[SAVE]**

To activate the relay contact

1. Lift handset or press **[SPEAKER]** button.
2. Dial Door Open code, '#*1', '#*2', '#*3' or '#*4'.
3. Hang-up to return to idle.

Or,

1. Lift handset or press **[SPEAKER]**.
2. Press the **{DOOR OPEN}** button
3. Hang-up to return to idle.

Conditions

1. The number of relay contacts available varies based on the system size. Refer to Table 1.2-1 for the number of available contacts.
2. The Door Open feature dial code is based on the assigned External Contact as below
   External Contact 1 = '#*1'
   External Contact 2 = '#*2'
   External Contact 3 = '#*3'

External Contact 4 = '#*4'

3. The contacts are rated at 1 amp, 24 VDC.
4. A station will receive error tone if not allowed access to Door Open.(PGM 113-Btn 25)

Programming

SYSTEM	1. External Contact Control (PGM 168)
	2. Door Open Timer (PGM 181-Btn 5)
STATION	1. Door Open (PGM 113-BTN 25)

Related Features

LBC (Loud Bell Control)

Hardware

External Control Contact connected to a door-lock release mechanism.

2.35 EMERGENCY CALL E-911 (CALLER LOCATION) SUPPORT (NA –AUSTRALIA)

Description

The Emergency Call E-911 (caller location) Support feature integrates the iPECS to a PBX ANI Link unit, such as from Tone Commander, to provide caller id and location information to an emergency center. When an emergency call is placed from a station of the iPECS, an assigned emergency CO Line is used to connect the ANI link unit. The iPECS sends the station number, four (4) digits, to the unit as DTMF digits. The ANI Link unit places a call to the emergency center over a connected CAMA trunk. Caller Id and location information from the ANI Link unit database is sent to the emergency center in the proper format and the station is connected to the call.

This feature also provides a means to identify a Power Failure signal from the ANI Link unit. Typically, an ANI Link unit will incorporate a normally open contact that closes when power to the unit fails. If a Power Failure signal is detected, iPECS routes emergency calls via PSTN/ISDN circuits other than those marked for emergency call use until the Power Fail signal is removed.

Operation

Operation of this feature is automatic when assigned

Programming

STATION	1. Emergency CO/Group (PGM 112-Btn 18)
	2. Proctor Monitor (PGM 113-Btn 22)
CO/IP LINES	1. Proctor On/Off (PGM 141-Btn 12)
TABLES	1 Emergency Code Table (PGM 226)

Conditions

1. The system Loop Start lines connected to the ANI Link unit must be assigned to a separate CO line group and the individual CO/IP lines assigned for emergency service.
2. The ANI Link unit must be properly connected to an SLT port and an LGCM port on the same LAN as the MFIM. In addition, the ANI Link device must be co-located with the iPECS system, in the same building.
3. The ANI Link unit must be properly installed and connected to a CAMA trunk from the telco to send location information to the local emergency center.
4. It is the Customer's responsibility to maintain the ANI Link unit number/location database; there is no integration provided between the iPECS database and the ANI Link unit.
5. The system must be programmed for a three or four digit numbering plan. The system must send four digits to the ANI unit. When a three digit plan is used, a leading '0' is automatically added to the station number.
6. During a power failure, if the ANI Link unit Power Fail contacts, normally open, are connected to a system SLT port, the iPECS will recognize the fault and emergency calls are not sent to the ANI Link unit. In this situation, emergency calls are sent over normal system PSTN/ISDN circuits and the station digits are not sent.
7. SMDR output will identify any call made via the Emergency call feature, including the success or failure of calls.

Related Features

Emergency Call
Emergency Call Attendant Alert

Hardware

Tone Commander PBX ANI Link unit or equal
CAMA telco trunk
SLTM port connected to ANI Link unit
LGCM port connected to ANI Link unit

2.36 EMERGENCY CALL

Description

Regardless of a station's dialing restrictions (COS), the user may dial assigned Emergency numbers.

Operation

System

The system will automatically override any toll restrictions and process an assigned Emergency number call.

Conditions

Programming

STATION 1. Emergency CO/Group (PGM 112, BTN 18)

TABLES 1 Emergency Code Table (PGM 226)

Related Features

Hardware

2.37 EXECUTIVE/SECRETARY FORWARD

Description

iPECS Phones can be assigned as Executive/Secretary pairs. By activating DND, the Executive also activates Unconditional Call Forward to the Secretary, which will forward Executive calls to the Secretary. With the "CO Call to Secretary" option enabled, all CO calls to the Executive forward to the Secretary regardless of the Executive's station status. In addition, if the Secretary is in DND, Executive calls sent to the Secretary route back to the Executive if the "Call Exec If Sec in DND" option is enabled.

Each Executive can be assigned a "Grade" (01, highest ~12, lowest). Executives with a higher grade can call lower grade Executives overriding the Executive/Secretary forward.

With the "Icm Call to Secretary" option(PGM229-Btn5) enabled, all internal calls to the Executive(except for calls from higher or same grade executive) forward to the Secretary regardless of the Executive's station status.

Callers to an Executive in DND can leave a Message Wait indication. The message waiting indication is given to the Executive or Secretary station based on the system database and User Programming.

Operation

iPECS Phone

To activate/deactivate Executive/Secretary forward from the Executive iPECS Phone

1. Press the **[DND]** button to toggle Executive/Secretary Forward.

Conditions

1. The available number of Executive/Secretary pairs is provided in Table 1.2-1.
2. An Executive may have multiple Secretaries and a Secretary may have multiple Executives. Each forms a separate Executive/Secretary pair.

3. If the Secretary is busy when a call is received for the Executive, the caller will receive busy tone.
4. The Secretary may override the DND status of the Executive to Camp-On and transfer calls to the Executive.
5. A chain can be constructed by assigning the Secretary of one pair as an Executive of another. Although a chain may be constructed, a loop back is not allowed.
6. If an Executive has multiple Secretaries, calls will automatically route to the Executive's first idle Secretary.
7. The Executive may use Call Forward to send calls to stations other than the Secretary.
8. Messages are left at the Executive or Secretary based on assignment of "Left Msg Exec". When enabled (On), messages are left at the Executive's station.
9. The Executive Grade can be assigned only for Country Code '82', Korea.

Programming

STATION

1. DND (PGM 111-Btn 3)
2. Left Msg Exec (PGM 113-Btn 10)

TABLES

1. Executive/Secretary Pair (PGM 229-Btn 1)
2. CO Call to Secretary (PGM 229-Btn 2)
3. Call Exec If Secretary in DND (PGM 229-Btn 3)
4. Exec Grade (PGM 229-Btn 4))
5. Icm Call to Secretary (PGM 229-Btn 5)

Related Features

DND (Do Not Disturb)
Call Forward
Call Waiting/Camp-On
CLI Message Wait
Message Wait/Call Back
Call Transfer

Hardware

iPECS Phone

2.38 EXTERNAL AUTO ATTENDANT/VOICE MAIL

2.38.1 AA/VM Group

Description

The system provides support for an adjunct Auto Attendant/Voice Mail system via connection to SLT ports. When a call arrives for the External AA/VM Group, the system will search the group for an idle port and deliver the call.

Signaling information between the iPECS and AA/VM system may be assigned for in-band DTMF signaling or the SMDI (Simplified Message Desk Interface) signaling protocol over the assigned iPECS system RS-232 port.

Operation

System

The system will interface with the External AA/VM based on database assignments.

Conditions

1. Selection of SMDI or in-band signaling can be alternated using the 2nd Virtual Dip switch.
2. Only one AA/VM Group (External, VMIM/VSF or Feature Server) can be defined in the system. Multiple definitions may cause erroneous system operation.

Programming

STATION GROUPS	1. Station Group Assignment (PGM 190)
	2. Station Group Attributes (PGM 191)
TABLES	1. Voice Mail Dialing Table (PGM 234)
DIP SWITCH	1. Virtual Dip Switch (PGM 453-Btn 2)

Related Features

In-band (DTMF) Signaling
SMDI (Simplified Msg Desk Interface)
Auto Call Recording
Two-Way Record
VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

External AA/VM system

2.38.2 In-band (DTMF) Signaling

Description

The system may employ in-band signaling to communicate with an External AA/VM system. When a call is routed to the AA/VM SLT port, the system will send DTMF signals informing the AA/VM of the characteristics of the call. DTMF digit strings are assigned to various functions allowing the AA/VM to respond appropriately to the call. These definitions are entered in the "Voice Mail Dialing Table"

Operation

System

The system will interface with the External AA/VM based on database assignments.

Conditions

1. Selection of SMDI or in-band signaling can be alternated using the 2nd Virtual Dip switch.
2. Only one AA/VM Group (External, VMIM/VSF or Feature Server) can be defined in the system. Multiple definitions may cause erroneous system operation.

Programming

STATION GROUPS	1. Station Group Assignment (PGM 190) 2. Station Group Attributes (PGM 191)
TABLES	1. Voice Mail Dialing Table (PGM 234)
DIP SWITCH	1. Virtual Dip Switch (PGM 453-Btn 2)

Related Features

AA/VM Group
SMDI (Simplified Msg Desk Interface)
Auto Call Recording
Two-Way Record

Hardware

External AA/VM system

2.38.3 SMDI (Simplified Msg Desk Interface)

Description

The system may employ SMDI (Simplified Message Desk Interface) protocol to communicate with an adjunct AA/VM system. When a call is routed to an AA/VM SLT port, the system will send SMDI messages over the assigned SMDI RS-232 port, informing the AA/VM of the characteristics of the call.

SMDI Protocol

There are three types of SMDI messages. Within each message is an "Action Code", which defines the function or required action of the AA/VM system. Fields within the messages also define the called/calling station and station status. The various message types and definition of the fields are shown in the chart below.

Type I message

cr lf **MD** ggg mmmm a xxxxxxxxx sp yyyyyyyyy sp cr lf^Y

Type II message

cr lf **MD** ggg mmmm a xxxxxxxxx sp sp cr lf^Y

Type III message

cr lf **MD** ggg mmmm a sp yyyyyyyyy sp cr lf^Y

Field	Description
cr	carriage return
lf	line feed
MD	Message Desk
ggg	Message desk number, AA/VM system, default 001
mmmm	Message Desk terminal, 0001-9999, VM port
a	action code
xxx...x	called station number or station calling the VM group
yy…y	calling station number
sp	ASCII space character
^Y	end of SMDI message, control + Y (0x19)

The following table provides detailed information on the meaning and function of the various SMDI messages used.

Table 2.38.3-1 SMDI Messages

Action Code	Reason	Purpose	In-band code	Message Type	SMDI Message MD 001 0001-
A	Unconditional forward to VM	Put Mail	P#	II	A xxxxx yyyyy
B	Called Station busy	Busy Mail	P#3P	II	B xxxxx yyyyy
C	Disconnect, connected party	disconnect	*****	II	C xxxxx yyyyy
D	Direct Fwd to VM group	Get Mail	P##	II	D xxxxx yyyyy
E	Error, invalid number	Error	P#*5P	II	E xxxxx yyyyy
H	Two-way Record	record	None	II	H xxxxx yyyyy
I	DND	DND	P#*6P	II	I xxxxx yyyyy
N	No Answer	No Answer	P#*4P	II	N xxxxx yyyyy
R	Direct CO/IP ring to VM group	AA	None	III	R xxxxx yyyyy

Operation

System

The system will interface with the External AA/VM based on database assignments.

Conditions

1. Selection of SMDI or in-band signaling can be alternated using the 2nd Virtual Dip switch.
2. Only one AA/VM Group (External, VSF, VMIM or Feature Server) can be defined in the system. Multiple definitions may cause erroneous system operation.

Programming

STATION GROUPS	1. Station Group Assignment (PGM 190) 2. Station Group Attributes (PGM 191)
TABLES	1. Voice Mail Dialing Table (PGM 234)
DIP SWITCH	1. Virtual Dip Switch (PGM 453-Btn 2)

Related Features

AA/VM Group
In-band (DTMF) Signaling
Auto Call Recording
Two-Way Record
VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

External AA/VM system

2.38.4 Voice Mail Box Translation

Description

Stations can be assigned to employ a Voice mailbox number that is different from the station number. When a call is routed to voice mail, the system will identify the mailbox as the assigned VM box number in place of the station number. The VM box number can be any 4-digit number ('0001' to '9999'). An entry of 0000 or blank sends the station number to the VM system.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

1. VM box translation applies to Feature Server and Adjunct In-band and SMDI Voice mail systems.

Programming

STATION	1. Voice Mail Id (PGM 113-Btn 14)
STATION GROUPS	1. Station Group Assignment (PGM 190)
	2. Station Group Attributes (PGM 191)
TABLES	1. Voice Mail Dialing Table (PGM 234)

Related Features

AA/VM Group
SMDI (Simplified Msg Desk Interface)
Two-Way Record
VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

2.39 FLEXIBLE NUMBERING PLAN

Description

User access to the iPECS system resources and features is accomplished through feature codes or iPECS Phone buttons. The Administrator selects from one of the eight (8) different standard numbering plans and, if desired, assigns codes for individual functions in the Flexible Numbering Plan. The feature codes are defined in the system's Flexible Numbering Plan; refer to Appendix B of the iPECS Admin and Programming Manual.

Operation

System

System implements feature activation based on the Flexible Numbering Plan.

Conditions

1. Feature codes can be one to three digits in length.
2. During programming, conflicts in the Numbering Plan are not allowed. The existing non-conflicting Numbering Plan is used until correctly updated.

Programming

SYSTEM ID	1. System Id (PGM 100-Btn 4)
NUMBERING	1. Flexible Station Numbering Plan (PGM 105)
	2. Flexible Numbering Plan (PGM 106 – 109)

Related Features

Hardware

2.40 8 DIGITS

Description

We can support it more 4 digits for station number. Station numbering should not conflict with numbering plan. It is consist of prefix digits and add digits. Prefix digits can have up to 4 digits and Add digits can have up ot 4 digits. If you set Prefix digit use ON, you should consider it interface with application, TAPI and Ez-Attendant. If you don't have TAPI and Ez-Atd that support 8 digits for iPECS-LIK system, it didn't work.

Operation

System

Operation of 8 digits should be programmed in PGM 100 and 238

Conditions

1. PGM 100-B5 : Prefix Usage
   If you set ON, It's influenced all admin and operation for interface with 3rd Party.
2. PGM 238 : 8-digit tables
   1) Prefix digits : 81 , Add Digits : 3 ➔ 5 digits use : ex) 81xxx
   2) Prefix digits : 8 , Add Digits : 4 ➔ 5 digits use : ex) 8xxxx
   3) Prefix digits : 2345 , Add Digits : 4 ➔ 8 digits use : ex) 2345xxxx

   Prefix digits cannot be conflicted with other numbering plan.

Programming

1. Prefix usage ON/OFF(PGM 100)
2. Prefix digits (PGM 238)

Related Features

Station Numbering Plan

Hardware

2.41 HEADSET COMPATIBILITY

Description

An industry standard headset can be connected to an iPECS Phone in place of or in addition to the handset. The station is then programmed for Headset operation.

In the Headset mode, pressing the **[SPEAKER]** button will send audio to the Headset instead of the speakerphone. In addition, when in the Headset mode, ring signals can be delivered to the speaker or the headset as defined in the system database.

iPECS Phone users may assign a Flex button to select Headset or Speakerphone operation. The **{HEADSET SELECT}** button may be used to toggle the operation of the phone between Headset and Speakerphone.

Operation

iPECS Phone

*To assign a Flex button for **{HEADSET SELECT}***

[PGM] + **{FLEX}** + **[PGM]** + '61' + **[SAVE]**

To change operation from Speakerphone to Headset

1. Press the **[PGM]** button.
2. Dial '61', the Headset select code.
3. Dial '0' to select Headset, '1' to select Speakerphone.

Or,

1. Press **{HEADSET SELECT}**.

To change the device to receive ring signals

1. Press the **[PGM]** button.
2. Dial '62', the Ring select code.
3. Dial '1' for Speaker, '2' for Headset or '3' for both.

To place/answer calls using the headset

1. Press the **[SPEAKER]** with the phone in Headset mode.

Conditions

1. The Intercom Signaling Mode can be set in the Headset mode as with the Speakerphone mode.
2. The station always receives Page announcements over the speaker of the iPECS Phone.
3. Although the phone is in the headset mode, the system will monitor the hook-switch status. If the user lifts the handset to go off-hook, audio is delivered to the handset.

Programming

STATION

1. Speaker/Headset Ring Mode (PGM 111-Btn 8)
2. Speakerphone/Headset (PGM 111-Btn 9)

Related Features

Speakerphone
Paging

Hardware

2.42 HOLD

2.42.1 Hold Preference

Description

Hold Preference defines either Exclusive Hold or System Hold as the preferred hold state, which is activated on the first depression of the **[HOLD]** button. The selection is based on the assigned Hold Preference in the system database.

Operation

iPECS Phone

To activate the Preferred Hold while on a call

1. Press the **[HOLD]** button, the connected party is placed in the preferred hold state.

Conditions

1. A transferred call is placed in the Exclusive hold state at the receiving station regardless of the assigned Hold Preference.
2. Hold Preference is not available to an SLT or non-iPECS VoIP terminals.

Programming

SYSTEM 1. Hold Preference (PGM 160-Btn 7)

Related Features

Call Transfer, CO/IP
Exclusive Hold
System Hold

Hardware

iPECS Phone

2.42.2 Hold Recall

Description

When a user places a CO/IP call on hold, a hold timer is activated. If the timer expires, the held call will recall the station for the I-Hold Recall time. If the call remains unanswered, the Attendant also receives recall for the Attendant Recall time and, if the call is on Exclusive hold, the call is placed on System Hold. If still unanswered after the Attendant Recall time, the CO/IP call is disconnected and the appropriate circuits returned to idle.

Operation

Hold Recall operation is automatic

Conditions

1. Separate timers are assigned for the various types of hold: System, Exclusive, Transfer, etc.
2. If the I-Hold timer is set to zero, the station will not receive recall. If the Attendant Recall timer is set to zero, the Attendant will not receive recall.
3. If the specific Hold timer is set to zero, recall is disabled.

Programming

SYSTEM

1. Hold Preference (PGM 160-Btn 7)
2. Attendant Recall Timer (PGM 180-Btn 1)
3. Exclusive Hold Recall Timer (PGM 180-Btn 4)
4. I-Hold Recall Timer (PGM 180-Btn 5)
5. System Hold Recall Timer (PGM 180-Btn 6)
6. Transfer Hold Recall Timer (PGM 180-Btn 7)

Related Features

Call Transfer, CO/IP
Exclusive Hold
System Hold

Hardware

2.42.3 Exclusive Hold

Description

CO/IP lines may be placed in a waiting state such that other stations in the system are unable to access the CO/IP line. Only the station placing the line on Exclusive hold can access the held line.

If the call remains on hold at expiration of the Exclusive Hold Recall Timer, normal Hold Recall will apply.

Operation

iPECS Phone

To place a call on Exclusive Hold

1. Press the **[HOLD]** button twice.

To access a call on Exclusive Hold from the holding station

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the held CO/IP line.

Or,

1. Dial ‘8#’, the Held CO/IP Call Access code.
2. Dial the CO/IP line number.

SLT

To place a call on Exclusive Hold

1. Momentarily press the hook-switch.

To access a call on Exclusive Hold from the holding SLT

1. Lift the handset.
2. Dial ‘8#’, the Held CO/IP Call Access code.
3. Dial the CO/IP line number.

Conditions

1. When a CO/IP line is placed on Exclusive Hold, the button LED will flash at 120 ipm at the holding station and will be On at all other stations.
2. All Transferred CO/IP calls are placed on Exclusive Hold for the receiving station.
3. The LED of **{LOOP}** and **{POOL}** buttons will display the CO/IP line status.

Programming

SYSTEM

1. Hold Preference (PGM 160-Btn 7)
2. Attendant Recall Timer (PGM 180-Btn 1)
3. Exclusive Hold Recall Timer (PGM 180-Btn 4)
4. I-Hold Recall Timer (PGM 180-Btn 5)
5. Transfer Hold Recall Timer (PGM 180-Btn 7)

Related Features

Call Transfer, CO/IP
Hold Preference

Hold Recall
System Hold

Hardware

2.42.4 System Hold

Description

CO/IP lines may be placed in a waiting state such that other stations in the system are able to access the CO/IP line. Stations must have access to the CO/IP line in the system database to access the held call.

If the call remains on hold at expiration of the System Hold Recall Timer, normal Hold Recall will apply.

Operation

iPECS Phone

To place a call on System Hold

1. Press the **[HOLD]** button.

To access a call from System Hold

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **{CO}**/**{IP}** line button

Or,

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial '8#', the Held CO/IP Call Access code.
3. Dial the CO/IP line number

SLT

To place a call on Exclusive Hold

1. Momentarily press the hook-switch.
2. Dial '560', the System Hold feature code.

To access a call from System Hold

1. Lift the handset.
2. Dial '8#', the Held CO/IP Call Access code.
3. Dial the CO/IP line number.

Conditions

1. When a CO/IP line is placed on System Hold, the button LED will flash at 30 ipm and wink at the holding station and will flash at all other stations.
2. A call on System Hold can be retrieved from any station allowed access to the CO/IP line in the system database using the CO/IP line button or the Held CO/IP call access code.
3. The LED of **{LOOP}** and **{POOL}** buttons will display the CO/IP line status.

Programming

SYSTEM

1. Hold Preference (PGM 160-Btn 7)
2. Attendant Recall Timer (PGM 180-Btn 1)
3. I-Hold Recall Timer (PGM 180-Btn 5)
4. System Hold Recall Timer (PGM 180-Btn 6)

Related Features

Call Transfer, CO/IP
Hold Preference
Hold Recall
Exclusive Hold

Hardware

2.42.5 Automatic Hold

Description

While on an active CO/IP call, the system will place the call on hold automatically if the user presses the **[FLASH]**, **[CONF]**, **{DSS/BLF}** or other feature buttons. In addition, the station can be programmed to support CO/IP to CO/IP Automatic Hold. In this case, pressing a CO/IP button while on a CO/IP call will place the active call on hold and access the selected CO/IP line.

Operation

iPECS Phone

To use Automatic Hold while on an active CO/IP call

1. Press the desired feature button or **{CO}**/**{IP}**; the active call is placed in the Preferred Hold state.

Conditions

1. CO/IP lines placed on hold with Automatic Hold are placed in the assigned Preferred Hold.
2. There is no limit on the number of calls that can be placed on hold using Automatic Hold.

Programming

STATION 1. Automatic Hold (PGM 112-Btn 2)
SYSTEM 1. Hold Preference (PGM 160-Btn 7)

Related Features

Hold Preference
Hold Recall

Hardware

iPECS Phone

2.43 HOT DESK

Description

iPECS Phones can be assigned as Hot Desk phones allowing users (Agents) to login with the iPECS. The Hot Desk will become active and will take on the attributes defined for the Agent's station number. When the Agent logs off, the Hot Desk phone becomes inactive and the Agent's calls are forward to the user-entered destination. A different Agent may then login through the inactive Hot Desk phone.

Operation

iPECS Phone

To login through an inactive Hot Desk;

1. Lift the handset or press the **[SPEAKER]** button,
2. Dial '*0',
3. Dial the user's station number and authorization code, Agent is logged in.

To logout through the active Hot Desk;

1. Press the **[PGM]** button,
2. Dial '* *' or press the **{AGENT LOGOUT}** Flex button,
3. Use the **[VOL UP]/[VOL DOWN]** buttons to select call forward destination for Agent calls (speed dial, joined mobile phone, VMIM/VSF, or VM group).
4. Press the **[SAVE]** button.

Conditions

1. The Hot Desk phone can be programmed to log-out an active user automatically if no action has been taken by the Agent for the Auto Log-out timer.
2. An active (logged in) Agent can login to another inactive Hot Desk. However, this will log the Agent off the previously active Hot Desk.
3. A user may only logout from the Agent's active Hot Desk phone. Attempting to logout using a different Hot Desk phone will return error tone.
4. The Flex button map of the Hot Desk station is fixed and does not take on the configuration associated with the Agent's station.
5. When logged off, the Agent's active database, including the following items, are saved.

Station Number
Station Attributes, PGM 111~124
CO Routing, Ring assignments, DID routing, etc.
Voice Mail
Station Group assignment

6. The number of Hot Desk phones and Hot Desk users is limited to the system capacity. Each Hot Desk phone and Hot Desk user (Agent) requires a separate station channel in the system.
7. To identify the station number, only station authorization codes may be used for Hot Desk login.
8. An ACD Agent must use the Agent Login procedure as defined under the ACD feature description.
9. DKT Phone cannot be Hot Desk.

Programming

STATION 1. Hot Desk (PGM 112-Btn 13)

TABLES 1. Hot Desk Attributes (PGM 250-Btn 1~3)
2. Auto Logout Timer (PGM 250-Btn 3)

Related Features

Station Groups
Automatic Call Distribution
Call Forward

Hardware

iPECS Phone

2.44 ICLID CALL ROUTING

Description

The system can employ ICLID (Incoming Calling Line Id) to determine the routing of incoming external calls. Each CO/IP Line, including DID Lines, may be assigned to employ ICLID routing. The system will compare the received ICLID to entries in the ICLID Routing Table and, if a match is found, will route the call to the destination defined in the ICLID Ring Assignment Table. Destinations can be the VMIM/VSF, an external Voice Mail, a station or a station group.

An ACD group may be assigned to route calls employing the ICLID Tables. When configured, calls re-route based on the Caller Entered ICLID.

Operation

System

System implements routing automatically based on database entries and the received ICLID.

Conditions

1. If the received ICLID does not match an entry in the ICLID Route Table, the call is routed based on the type and other programming (Ring assignments, etc.) for CO/IP Line.
2. For analog CO Lines, the system will await receipt of valid ICLID for the ICLID Ring Timer. At expiration of the timer, if ICLID is not received, the call is routed based on the type and other programming (Ring assignments, etc.).
3. The ICLID received from the CO/IP Line must be a telephone number to match an ICLID Route Table entry.
4. If ICLID routing is enabled for a DID line, DID Call Wait is disabled.

Programming

CO/IP	1. ICLID Ring Timer (PGM 142-Btn 14)
ISDN/ICLID	1. ICLID Route Table (PGM 203) 2. ICLID Ring Assignment Table (PGM 204
STATION GROUPS	1. ACD ICLID Use (PGM 191-Btn 21)

Related Features

CO/IP Ring Assignment
Automatic Call Distribution
Direct Inward Dial (DID)

Hardware

2.45 IN-ROOM INDICATION

Description

A supervisor can press the In-Room Indication button and **[HOLD/SAVE]** button when in an idle state; then each Station's In-Room Indication LED will illuminate.

10 groups can be programmed. Each group has at most 20 members excluding Supervisor.

Operation

To program a flexible button as the "In-Room Indication button"

1. Press the **[TRANS/PGM]** button and the flexible button to be assigned.
2. Dial the code (TRANS/PGM + 9* + Room NO(bin no))
3. Press the **[HOLD/SAVE]** button.

To Active or Deactivate In-Room Indication button,

1. Check if Supervisor station is the idle state and In-Room Indication button is programmed.
2. Press In-Room Indication button and press **[HOLD/SAVE]** button.

Conditions

5. If Supervisor station is not Idle, In-Room Indication button does not work.

- If a station press the In-Room Indication button but the station is not supervisor, an error tone is heard.
- If **[HOLD/SAVE]** button is not pressed within 5 seconds after In-Room Indication button pressed, the station goes back to the Idle state.

Programming

SYSTEM 1. In-Room Indication (PGM 183)

Related Features

Hardware

2.46 IP BRIDGE MODE

Description

The Phontage and UCS client can automatically bridge the audio for a call to linked pair station. When the Phontage or UCS client place a call in the bridged mode, the iPECS Phone is connected to the call.

Operation

To use the IP Bridge mode:

1. Place a call from the Phontage or UCS client. The Phontage or UCS client must be in the Bridged mode.
2. The call must be connected by lifting the handset of the linked iPECS Phone.

Conditions

1. The Phontage or UCS client must be used locally to the bridged iPECS Phone
2. Bridged operation must be selected at the Phontage or UCS client.
3. The Phontage or UCS Phone must be linked to the associated iPECS Phone.
4. Bridging is not available from the iPECS Phone to the Phontage or UCS client.

Programming

Related features

Linked Station Pairs

Hardware

2.47 IP System DECT

Description

iPECS supports office building mobility employing Digital European Cordless Technology (DECT). LG-Ericsson's DECT Base stations (GDC-400B) connect to the Wireless Telephone Interface Module (WTIM). The WTIM manages up to eight (8) base stations and up to 3 WTIMs may be linked to provide a DECT coverage zone with transparent handover. Within the Zone, DECT handsets (GDC-400H) can roam and maintain an uninterrupted communications link to iPECS features and resources through the base station to the WTIM. Additional zones can be created to the maximum number of WTIMs allowed for the system, see Table 1.2-1.

For further information on installation and operation of the IP System DECT solution, refer to the IP System DECT Manual.

Operation

DECT operation is automatic when configured.

Conditions

1. Multiple, up to three (3) WTIMs may be linked to provide a zone with up to 24 base stations, (8 per WTIM).
2. The maximum number of WTIMs that can be installed in an iPECS system varies based on the system model, refer to Table 1.2-1.

Programming

DECT

1. DECT Registration (PGM 491-Web only).
2. DECT Attributes (PGM 491)

Related features

Hardware

WTIMs
GDC-400B Base stations
GDC-400H handsets

2.48 IP FAX RELAY, T.38 SUPPORT

Description

Because of there nature, Fax tones do not transmit well through IP networks, particularly when compression is employed. To address this, the SLTM, LGCM, BRI and PRI gateways support T.38 protocol that defines the translation of fax tones to digital signals. When Fax tone is detected on a port of an iPECS module, the module will activate a T.38 Fax relay channel to the appropriate Line and SLT module.

Operation

Operation of this feature is automatic.

Conditions

Programming

Related Features

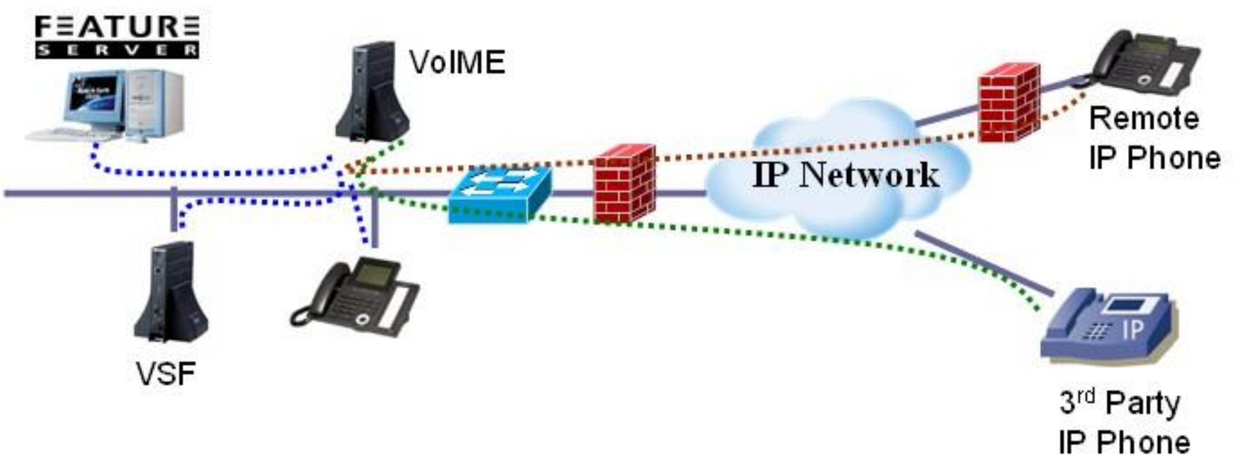
Hardware

2.49 IP TRANS-CODING

Description

The system employs either IEEE g.711, g.729 or g.723 codec to digitize and compress voice signals for RTP packets between iPECS devices. iPECS terminals and gateway Modules incorporate DSP functions to support codec conversion Available VOIMs include DSP circuitry used to support trans-coding (converting) codecs for incoming VoIP calls to devices, such as the MFIM VoIP channels, the

VSF channels and the Feature Server, which have no built-in codec. The VOIMs will trans-code the incoming voice codec (g.711, g.723, g.729) to the system codec and reverse the process for the outgoing packets. When the external VoIP connection can only support g.729 and the system codec is g.723, the DSP must implement a complex trans-coding operation, which requires 2 DSP channels. In all other cases, trans-coding only requires a single channel per call.



Operation

System

System flow with trans coding:

Conditions

1. The system codec toward the VOIM can be changed anytime within an IP call.
2. The VOIM DSP can generate and detect in-band DTMF and Call Progress tones in support of DISA functionality.
3. For complex trans-coding (g.723/g.729), the VOIM DSP will require 2 channels.
4. If there are no available channels when trans-coding is required, the VOIM gateway will release call control.

Programming

Related Features

System Networking
Remote Device Zone Management

Hardware

VOIM8 or VOIM24

2.50 LNR (LAST NUMBER REDIAL)

Description

The last number dialed is stored (up to 48 digits) in the station's Last Number Redial buffer. The user may request the system redial the last dialed number without the need to dial the number.

For iPECS Phones with displays, the last 10 numbers are stored in the LNR buffer. The user may view the numbers using the **[VOL UP]/[VOL DOWN]** button and select the number to dial from the list.

Operation

iPECS Phone

To use Last Number Redial using the [Redial] button;

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[REDIAL]** button.
3. Press the **[VOL UP]/[VOL DOWN]** button to highlight the desired number.
4. Press **[SAVE]** or **[REDIAL]** to dial the number highlighted.

To use Last Number Redial using the LIP-8000 series soft buttons

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Dial '*'.
4. Use the **Navigation** button to select the desired number.
5. Press the **[HOLD/SAVE]** button to place the call.

SLT

To use Last Number Redial

1. Lift the handset.
2. Dial '552', the Last Number Redial code.

Conditions

1. For iPECS Phones with display, the redial buffer will store duplicate numbers unless dialed consecutively.
2. When the CO/IP line used for the original call is busy, the system will select an idle line from the same CO/IP line Group to place the call.
3. Using Last Number Redial will cancel Automatic Called Number Redial if active.
4. The LNR buffer is not stored in non-volatile memory and is erased if power to the system is lost.
5. Manually dialing a Flash during an outgoing call will cause only those digits dialed after the Flash to be stored in the LNR buffer.
6. LNR applies to both CO and VoIP calls.

Programming

Related Features

Save Number Redial (SNR)
Station Speed Dial
System Speed Dial

Hardware

2.51 LCR (LEAST COST ROUTING)

Description

The LCR Tables are employed to define appropriate routing for outgoing calls based on the dialed number. Generally, the LCR Tables are structured to define the Least Cost Route for Long Distance calls.

User dialed digits are compared to table entries and modified based on time of day, day of week, and assigned routes. There are four LCR Tables: LCR Control Attributes, LCR Leading Digit Tables, LCR Digit Modification and LCR Initialization Tables.

LCR Access modes are assigned in the LCR Control Attributes Table. These modes define the manner in which the user accesses the LCR function. LCR may be disabled or one or more of the three access modes can be allowed to access LCR. The basic modes are:

1. Internal LCR

If the user dialed digits match an Internal LCR code in the LDT (Leading Digit Table), a CO/IP path is selected and digits are modified using the DMT (Digit Modification Table).

2. Loop LCR

If the user dials 9 (the 1st available CO line access code) or presses a **{POOL}** or **{LOOP}** button to place a call, the Loop LCR mode is accessed. Dialed digits are compared to the Loop LCR codes in the LDT; the system will seize a CO/IP line from the assigned CO/IP Group and sends digits from the DMT.

3. Direct CO LCR

If the user selects a **{CO}**/**{IP}** or **{CO/IP GROUP}** button, the Direct CO LCR mode is accessed. If the user-dialed number matches a Direct LCR code in the LDT and the seized CO/IP line belongs to the assigned Direct CO LCR group, the system will send digits modified based on the DMT.

Operation

System
Operation of LCR is automatic based on assignments in the system's LCR Tables

Conditions

1. There are a total of 6 LCR access mode combinations that can be defined as below:
    1) LCR Access Mode 00 (M00) – LCR is disabled.
    2) LCR Access Mode 01 (M01) – Loop LCR access is active.
    3) LCR Access Mode 02 (M02) – Loop and Internal LCR access are active.
    4) LCR Access Mode 11 (M11) – Direct CO and Loop access are active
    5) LCR Access Mode 12 (M12) – Direct CO, Loop and Internal LCR access are active, seize Co line according to the LCR attributes
    6) LCR Access Mode 13 (M13) – Direct CO, Loop and Internal LCR access are active, seize CO line according to the station attributes

2. Multiple Leading digit table can be provided for each station, CO line.
3. Leading digit entries may be duplicated in the Leading Digit Table. Using a different "LDT Index" will make each entry unique. The system will use the lowest matching entry.
4. When Direct CO LCR is used on an ISDN line, an ISDN information message with the Called party Info Element, which includes only the numbering plan and numbering type, is sent to the ISDN to maintain the ISDN connection while the user finishes dialing and the system modifies the digits.
5. For Direct CO LCR, the number of digits for the LDT Table should be programmed considering the dial-tone time-out of the network.
6. Since a CO/IP path is connected, Direct CO LCR does not support the Alternative DMT index, which allows the system to select a second or alternative CO/IP path to place the call.
7. If the LCR CO group is not assigned, the system will not seize a CO line call and make internal call.
8. If the LCR CO group is assigned as the unused group, the system it will seize a CO line according to the station attributes.

Programming

TABLES

1. LCR Assignment (PGM 220)
2. LCR LDT Table (PGM 221)
3. LCR DMT Table (PGM 222)
4. LCR Table Initialization (PGM 223)

Related Features

CO/IP Access
CO/IP Line Groups
ISDN (Integrated Service Digital Network)
Station Flexible Buttons

Hardware

2.52 Linked Station Pairs

Description

A station can be logically linked to a primary station so that the two stations function as a single station. When linked, the two stations effectively act as a single station with the station attributes of the primary station. The status of one station is reflected in the status of the other and features activated at one are active at the other. All internal or external calls to a linked pair station will ring both stations.

All features available to the primary station are available and controllable by the secondary station, one station may activate Call Forward and the other may cancel the forward. The displays of the linked stations will display the status of the linked pair. When one is busy, the display of the linked station will be as shown below.

IN USE AT LINK STA

Operation

System

Operation of Linked pairs is automatic when defined

Conditions

1. Any combination of iPECS Phones and SLTs may be assigned as Linked pairs. However, a DSS Console may not be assigned as a linked pair station.
2. Intercom calls to the Linked stations always signal in the Tone ring mode and cannot be changed using the Caller Controlled ICM Signaling feature.
3. Linked pair stations are treated as having a single station number for all features including LCD displays, station programming, ADMIN access, ACD statistics, SMDR, etc.
4. The station attributes of the Secondary station will follow attributes of the Primary station, i.e. Day/Night COS, CO Warning Tone, CO Auto Hold, CO Call Drop, DID Call Waiting, Speed Access, Alarm, VSF Access, DND, FWD, Paging, CO Line Access, CO Ring Assign, etc.
5. If one station of a Linked pair is busy, the other station of the linked pair is also considered as busy, thus use of the linked station to place a call is not supported.
6. A station can be linked to another station without registration to the system. This allows a station to be linked without affecting the overall capacity of the system. In this case, only an iPECS phone, Phontage or SLT attached to an SLTM2 can be used as the unregistered linked

station. In other cases, the linked station must be registered with the system, reducing the system capacity by one.
7. Linked pair stations cannot connect with each other between Master and Slave but ring for indication.
8. Linked pair cannot transfer a call with each other. But if Master(or slave) transfer to Slave(or Master) and on hook, you can recall and hold the call in slave(or Master).

Programming

SYSTEM 1. Linked Station Pairs (PGM 124)

Related Features

Intercom Caller Controlled ICM Signaling

Hardware

2.53 LBC (LOUD BELL CONTROL)

Description

The iPECS hardware is equipped with relays that activate External Control Contacts. Each contact is assigned to one of several functions including a Loud Bell Control. If used as a Loud Bell Control, the contact will activate when:

External Page is accessed
Assigned Station receives ring (LBC)
UNA (Universal Night Answer) CO/IP line receives a call while in the Night or Timed Ring mode.

Operation

System

Operation of the relays is automatic when defined

Conditions

1. The number of available contacts is given in Table 1.2-1.
2. When assigned to activate as a Loud Bell Control, CO/IP ring, transfers, and tone ring Intercom calls to the assigned station will activate the contacts.
3. The contacts are rated at 1 amp, 24 VDC.
4. If the system is in the Night or Timed ring mode with External Night Ring enabled, a call on a CO/IP line assigned with UNA will activate the LBC 1 contact. This permits operation of an external night ring device. While in the Night or Timed ring mode, LBC1 is not activated by ringing at the assigned station.

Programming

CO/IP 1. Universal Night Answer (PGM 141-Btn 7)
SYSTEM 1. External Night Ring (PGM 160-Btn 6)

2. External Contact Control (PGM 168)

Related Features

Universal Night Answer
Door Open

Hardware

External Control Contact connected to an external loud bell.

2.54 MOBILE EXTENSION

Description

A mobile phone may be registered to a station allowing the mobile phone to place and receive calls through the system. ISDN DID calls are sent to the user's iPECS Phone and the active registered mobile phone simultaneously. If the mobile phone is paired with a Hunt group station, Hunt group calls routed to the station also ring to the active mobile phone when enabled.

The mobile phone users can access the facilities of the iPECS to place internal and external calls as well as activate/access features. To access system facilities and resources, the mobile user calls the DID number of the corresponding iPECS Phone. When the call is received, the system matches the CLI to the mobile phone and provides the mobile user with system dial tone.

The user may be allowed to register and activate a mobile phone.

Operation

iPECS Phone

To register a the mobile phone number;

1. Press the **[PGM]** button,
2. Dial '37'.
3. Dial the mobile phone number.
4. Press the **[SAVE]** button.

To activate a registered mobile phone from the user's station:

1. Press the **[PGM]** button.
2. Dial '38'.
3. Dial digit '1' to activate, '0' to deactivate.
4. Press the **[SAVE]** button.

To place a call from the mobile extension using the iPECS:

1. Dial the DID number of the station, the system will check the CLID, answer the call and the user will receive intercom dial tone.
2. Place internal or external iPECS call as normal.

To Transfer a call from the mobile extension using the iPECS:

1. Dial '*' while on an iPECS call.
2. Dial the desired extension, the call is transferred and the mobile phone returns to idle.

Note: the mobile may reconnect by dialing '#'.

System
Incoming DID calls are sent to active mobile phones automatically.

Conditions

1. When the mobile phone places an external call through iPECS, the CLI of the corresponding station is used.
2. The Mobile Extension features are supported via system digital (T1 and ISDN) lines only.
3. Message Wait and Callback cannot be activated to a mobile phone.
4. The Mobile Extension feature is not supported over a distributed networked environment.
5. When an incoming ISDN DID call is received, the system will access an ISDN line and place a call to the mobile phone. Thus, an ISDN line must be available for the system to notify the mobile user of the incoming call.
6. Hold and Transfer Recalls to the mobile phone are sent to mobile phone and the associated station.
7. ACD, Circular and Terminal Hunt Group calls can be routed to the active Mobile Extension.

Programming

SYSTEM 1. Mobile Extension Table (PGM 236)

Related Features

DND (Do Not Disturb)
Station Message Wait/Call Back
Attendant Recall
Distributed Control Network
Station User Programming & Codes

Hardware

iPECS Phone

2.55 MULTIPLE LANGUAGE SUPPORT

Description

With the VSF or VMIM, iPECS can support six (6) languages simultaneously. Prompts in the desired languages are loaded into the VMIM/VSF memory along with the Language Selection prompts. To assure the proper language is employed, the Language Selection prompts is played when an

incoming call is assigned to be answered by a DID, DISA, Auto Attd or station hunt group announcement. The Language Selection prompts are played in multiple phrases, one in each of the equipped languages, with a request for the caller to input a digit to select the appropriate language. The system then employs the defined announcement (DID, DISA, etc.) recorded for the selected language.

Operation

System

System automatically plays the Language Selection prompts and plays prompts in the selected language.

To record a VMIM/VSF Multi Language Selection prompts

1. Press the **[PGM]** button.
2. Dial '06', the VMIM/VSF Record code.
3. Dial the VMIM/VSF device Sequence number.
4. Dial the VMIM/VSF Multi Language selection Announcement number, 72
5. Dial the Language number, only required with multi-language support. The current announcement is played followed by the "Press # to record" prompt.
6. Dial '#'.
7. After the beep-tone, record message.
8. Press the **[SAVE]** button to stop recording and save the message.

Conditions

1. The Language Selection prompts must be recorded as announcement number 72.
2. Multi-language support is only available with the VMIM and VSF.
3. Separate announcements must be recorded by the Attendant for each language supported.

Programming

SYSTEM 1. Multi-language (PGM 179)

Related Features

VMIM/VSF Integrated Auto Attd/Voice Mail
Direct Inward Dial (DID)
Station Groups
Announcements
Direct Inward System Access (DISA)

Hardware

VSF and/or VMIM

2.56 MULTIPLE VOICE MAILBOX SUPPORT

Description

A station can access any Voice Mailbox by dialing the VM group, the mailbox number and password. iPECS Phone users may assign one or more Flex buttons to access a specific mailbox and receive a visual indication (LED flashes) of message status in the mailbox. A Flex **{VMAILBOX}** button is assigned to allow access to multiple Voice Mailboxes.

Operation

iPECS Phone

*To assign a **{VMAILBOX}** Flex button;*

[PGM] + **{FLEX}** + VM group + Mail-box (station) number + **[SAVE]**

*To access a Voice Mailbox using the **{VMAILBOX}** Flex button;*

1. Lift the handset or press the **[SPEAKER].**
2. Press the **{VMAILBOX}** Flex button.
3. Dial the Mailbox password.

Conditions

1. A Flex button can be assigned to access the Mailbox for a Station Group or CO/IP Line Voice Mailbox.
2. The **{VMAILBOX}** button LED will flash indicating new messages have been received in the associated mailbox.
3. The Flex button cannot be assigned to access the mailbox of a networked station.
4. An adjunct Voice Mail may not provide proper notification to the system of message status and thus, the **{VMAILBOX}** button LED may not properly indicate new messages.

Programming

Related Features

External Auto Attendant/Voice Mail
VMIM/VSF Voice Mail

Hardware

iPECS Phone
External or VMIM/VSF Voice Mail

2.57 MOH (MUSIC-ON-HOLD)

Description

When a call is placed in a hold state, the system will deliver audio from the defined MOH source. In this way, the connected user can determine that the connection is still active.

The system has connections for up to two music sources. The first, BGM1 can be either internal or external source connected to either of the BGM1 inputs. The second source (BGM2) requires connection of an external source. Either source, BGM1 or BGM2, can be assigned to provide MOH.

In addition, to the BGM1 & 2, a message recorded in the VMIM/VSF can be employed as MOH. The Attendant records the VMIM/VSF announcement for MOH and VSF MOH is assigned as the MOH source. Separate messages can be recorded for each of the 3 languages supported by the system. Also, SLTM Port can be employed as MOH. Maximum 5 SLTM port can be configured as a SLTM MOH.

Operation

System

Operation of MOH is automatic

To record a VMIM/VSF announcement for MOH

1. Press the **[PGM]** button.
2. Dial '06', the VMIM/VSF Record code.
3. Dial the VMIM/VSF device Sequence number.
4. Dial the VMIM/VSF MOH Announcement number, 71
5. Dial the Language number, only required with multi-language support. The current announcement is played followed by the "Press # to record" prompt.
6. Dial '#'.
7. After the beep-tone, record message.
8. Press the **[SAVE]** button to stop recording and save the message.

Conditions

1. The internal source is provided as MOH when MOH Source 0 is selected in the system database.
2. The BGM1 RCA jack on the front panel of the MFIM and the BGM1 input via the rear panel RJ-11 jack are common. The external music source should only be connected to one of the inputs.
3. Only VMIM/VSF announcement number 71 may be used for the MOH message.
4. The iPECS-50 only has a single MOH/BGM source input.

5. The iPECS-Micro only has a single virtual MOH/BGM source input.

Programming

CO LINE	1. MOH (PGM 142-Btn 6)
SYSTEM	1. MOH Type (PGM 171-Btn 2)
RSGM	1. MOH Type (PGM 434-Btn 1)

Related Features

Hold
Back Ground Music
Multiple Language Support

Hardware

External Music source connected to MFIM music source input.

2.58 NETWORK MANAGEMENT SYSTEM

Description

iPECS NMS is a Web-based application for monitoring and management of multiple iPECS systems using standard SNMP (Simple Network Management Protocol). iPECS NMS is an efficient convenient tool employing standards based protocols and a Web based architecture to permit administrators remote access to iPECS systems using any common Web browser. iPECS NMS monitors multiple systems displaying real-time detailed status information for the system devices and channels. iPECS NMS maintains a log of alarm and fault events defined by the administrator and can alert administrators of potential service affecting faults. In addition, call statistics are maintained and can be reported with various tables and graphs.

Operation

Operation of iPECS NMS is automatic once configured
Administrative operations are covered in the iPECS NMS Manual

Conditions

1. iPECS NMS is subject to the conditions outlined in the iPECS NMS Manual.

Programming

SYSTEM	1. SNMP Attributes (PGM 197-Web only)

Related Features

Diagnostic/Maintenance

Hardware

2.59 NETWORK SECURITY & PRIORITY

Description

Each iPECS device (gateway Module or Terminal) supports several security and priority protocols. iPECS devices incorporate a Web server, which will deliver the Module Web Admin pages to a standard browser or the Module Admin may be accessed in the Terminal mode through the RS-232 port of the Module. Characteristics that can be established are:

- IEEE 802.1p/Q, VLAN – sets Virtual LAN tag and priority for Ethernet frame
- Diffserv – sets Diffserv Code Point priority for IP packet
- IPSec – enables IPSec to establish IPSec tunnel and encryption of IP packet
- SRTP – enables Secure RTP for RTP packet payload using AES (Advanced Encryption Service).

Operation

System

Operation of Security and priority is automatic once configured

Conditions

1. For Web Admin, the password is encrypted using the LG-Ericsson Java Virtual Encryption plug-in. A Java Virtual Machine (MS or Sun) must be installed in the user's PC to support password encryption.
2. Security and priority characteristics can be set for all devices, local or remote.
3. SRTP support is available only with the Phase 4 or higher modules. The devices must be on the same LAN to support SRTP.
4. iPECS implementation of IPSec employs a proprietary Key exchange protocol from the MFIM to the iPECS device.
5. IEEE 802.1 VLAN priority and ID are set at each device using the device Web interface. Diffserv, IPSEC and SRTP treatments may also be set in the device via the local Web interface. For the MFIM, VLANs are configured via the maintenance interface of the RS-232 port.

Programming

BOARD	1. SRTP (PGM 132-Btn 4)
SYSTEM	1. Web Password Encryption (PGM161-Btn15)

Related Features

Hardware

2.60 PRE-DEFINED & CUSTOM TEXT DISPLAY MESSAGES

Description

A user can select a text message to be displayed on the LCD of a calling iPECS Phone with Display. When a user activates Text Display Messages, incoming intercom calls will signal the user with Muted ring in-place of normal ring and the LCD of the calling station will display the selected message. There are ten Pre-defined messages (01-10), ten system wide Custom messages and one user defined Custom message. Several of the ten Pre-defined messages allow for auxiliary information such as a time, date or number.

System level Custom Messages may be entered from the Attendant or Administrator's phone or via the Web Admin. The user's Custom Message is assigned from the user's phone as well as the Attendant or the Administrator.

iPECS Phone users may assign a Flex button as a **{DISPLAY MESSAGE}** button or may include the message code to act as a **{ONE-TOUCH DISPLAY MESSAGE}** button.

The Pre-defined messages are:

Message 01

LUNCH RETURN AT hh:mm

Message 02

ON VACATION RETURN AT DATE mm:dd

Message 03

OUT OF OFFICE RETURN AT TIME hh:mm

Message 04

OUT OF OFFICE RETURN AT DATE mm:dd

Message 05

OUT OF OFFICE RETURN UNKNOWN

Message 06

CALL (enter up to 17 digits)

Message 07

IN OFFICE STA xxxx

Message 08

IN MEETING RETURN AT TIME hh:mm

Message 09

AT HOME

Message 10

AT BRANCH OFFICE

Operation

iPECS Phone

To assign a Flex button for Display Messages

{DISPLAY MESSAGE} button:

[PGM] + **{FLEX}** + **[PGM]** + '51' + **[SAVE]**

{ONE-TOUCH DISPLAY MESSAGE} button:

[PGM] + **{FLEX}** + **[PGM]** + '51' + message number + **[SAVE]**

To activate Display Message

1. Press the **[PGM]** button.
2. Dial '51', the Display Message code.
3. To view available messages, press **[VOL UP]/[VOL DOWN]** button.
4. Dial the Message number ('00'-'20').
5. Dial auxiliary input, as desired.
6. Press the **[SAVE]** button.

Or,

1. Press the **{DISPLAY MESSAGE}** button.
2. To view available messages, press **[VOL UP]/[VOL DOWN]** button.
3. Dial the Message number ('00'-'20').
4. Dial auxiliary input, as desired.
5. Press the **[SAVE]** button.

Or,

1. Press the **{ONE-TOUCH DISPLAY MESSAGE}** button.
2. Dial auxiliary input, as desired.
3. Press the **[SAVE]** button.

To cancel an active Display Message

1. Press the flashing **[FWD]** button.

To define the User Custom Text Message (00)

1. Press the **[PGM]** button.
2. Dial '52', the Custom Message program code.
3. Enter the Message contents, up to 24 characters using 2 dial pad digits for each character.

4. Press the **[SAVE]** button, confirmation tone is received, the new User Custom Text Display Message is stored and the station returns to idle.

SLT

To activate a Display Message
1. Lift handset.
2. Dial '561', the SLT Programming code.
3. Dial '51', the Display Message code.
4. Dial Message number ('00'-'20').
5. Dial auxiliary data as desired.
6. Momentarily press the hook-switch, confirmation tone is received.

To cancel an active Display Message
1. Lift the handset.
2. Dial '559', the SLT Feature Cancel code.

To enter the User Custom Message (00)
1. Lift the handset
2. Dial '561', the SLT Programming code.
3. Dial '52', the Custom Message program code, confirmation is received.
4. Enter the Message contents, up to 24 characters.
5. Momentarily press the hook-switch, confirmation tone is received.

System Attendant

To activate Display Messages for other stations
1. Press the **[PGM]** button.
2. Dial '051', Attendant Display Message code.
3. Dial station range.
4. To view available messages, press **[VOL UP]/[VOL DOWN]** button.
5. Dial Message number ('00'-'20').
6. Dial auxiliary data as desired.
7. Press the **[SAVE]** button.

To cancel active Display Messages for other stations
1. Press the **[PGM]** button.
2. Dial '052', Attendant Display Message Cancel code.
3. Dial station range.
4. Press the **[SAVE]** button.

To enter a Custom Message
1. Press the **[PGM]** button.
2. Dial '053', the Custom Message program code.
3. Dial desired Message code ('11'-'20').
4. Enter the Message contents, up to 24 characters.

5. Press the **[SAVE]** button, confirmation tone is received, the new Custom Message is stored and the station returns to idle.

Conditions

1. Only the System Attendant or Administrator can assign the contents of Custom Messages 11-20.
2. The alphanumeric characters are displayed as they are entered; refer to Station Speed Dial for entry procedures.
3. Display Message is cancelled if the user activates DND or Call Forward.
4. Custom Text Display Messages and Display Message status are stored in non-volatile memory to protect against loss during power failure.
5. The calling station must be an iPECS or LG-Ericsson digital phone with display to receive the message.
6. SLTs are notified of an active Display Message with stutter dial tone. iPECS Phones will have a flashing **[FWD]** button when there is an active Text Display Message.
7. Activating a Text Display Message does not affect normal operation of the station.
8. Pre-defined Messages 01 to 04 and 06 to 08 permit the user to input auxiliary information such as time, date or number.
9. The Attendant station can activate a Text Display Messages for other stations. However, this feature is not available to an Attendant.

Programming

Related Features

Silent Text Message
DND (Do Not Disturb)
Call Forward
Speed Dial

Hardware

iPECS Phone required to receive Display Messages

2.60.1 DND operation in internal call to pre-selected station

Description

When a station is set Pre-selected Message and other station call to the pre-selected station, The pre-selected station can receive the ring of internal call or not, by **[P-MSG DND]** Admin.

If **[P-MSG DND]** Admin of the pre-selected station is **ON**, the pre-selected station doesn't receive the ring and the caller hears DND tone. It is same operation with DND except that the "pre-selected massage" is printed in LCD of caller.

Operation

[P-MSG DND] admin is off

1. Call to Station XXX and Station XXX was set Pre-Selected MSG feature.
2. Station XXX can get the ring.
3. Caller hears ring-back tone.

[P-MSG DND] admin is on

1. Call to Station XXX and Station XXX was set Pre-Selected MSG feature.
2. Station XXX doesn't receive the ring.
3. Caller hears DND tone.

Conditions

Programming

STATION 1. PMSG-DND (PGM 114-Btn 22)

Related Features

Station Speed Dial
System Speed Dial

Hardware

2.61 REDUNDANT SYSTEM PROCESSOR

Description

The iPECS system supports redundant processor operation. The redundant system processor is a hot standby processor. The LAN2 port of the master MFIM is connected directly to the LAN2 port of the standby MFIM. The standby MFIM monitors the active MFIM and takes over system control if the:

active MFIM power fails.
active MFIM is reset.
LAN1 connection to the active MFIM fails or is removed.
standby MFIM does not receive a polling message from the active MFIM for 1 minute.

If failure occurs, the standby MFIM will take over and maintain control of the system. When the original master recovers from failure, it functions in the standby processor mode. The active MFIM will maintain the system database for both the active and standby processor.

Operation

Operation of redundant processors is automatic when installed.

Conditions

1. The Master/Slave switch on the main MFIM must be set in the Master position and the back up MFIM must be set in the Slave position to match the system Admin settings.
2. All eight wires of the Cat 5 cable must be terminated on the RJ45 connectors.
3. It is not possible to use the VSF(Internal VM) and a VMIM(external or commercial VM) should be installed for CPU redundancy. If you set redundancy ON by 161-Btn 20, VSF goes to "OUT of Service".
4. Master and Slave can be active in case LAN2 cable is out of order or other reason. These dual active can be protected using IP watch timer(181-B14). If you set the timer as 1 sec, Master send ARP to Slave every one second and Slave can detect Master alive. In this case, Slave can be deactive LAN1 for protect dual active.

Programming

NUMBERING PLAN

1. LAN2 Master IP Address (PGM 102-Btn 16)
2. LAN2 Slave Address (PGM 102-Btn 17)

SYSTEM

1. CPU Redundancy (PGM 161-Btn 20)
2. Change Active MFIM by Power Fail (PGM 161-Btn 21)
3. IP WATCH timer (PGM 181-Btn 14)

Related Features

Hardware

Main and back-up MFIMs, see the iPECS Description and Installation Manual section 4.4.2.

2.62 REGISTRATION & FRACTIONAL MODULE TABLES

Description

Normally, local iPECS devices can register with the system only when the "Registration Switch", 3rd Dip switch on the MFIM, is On. In this state, the system will allow any local iPECS device to register, providing a convenient "plug & play" initial installation. After initial installation, the Registration Switch is placed in the Off position, disabling registration of additional local device.

To eliminate the potential for unintended device registration, particularly where multiple systems exist on the same LAN, the system can be programmed to allow local device registration employing MAC addresses. Using the defined MAC address registration, the system allows devices with matching MAC addresses to register regardless of the Registration Switch position.

Three tables are provided to enter MAC addresses. The MAC Registration Table (**PGM** 102) allows entry of two MAC address ranges. Any device with a MAC address in either of the ranges will be permitted to register with the system.

The second is the Fractional Module Table (**PGM** 235). This Table permits entry of five MAC addresses and the number of channels to be registered for each device. Entering the MAC address permits the device to register with the system regardless of the Registration Switch position. The number of channels available to the device is limited to the number of channels entered in the Table. This function is commonly used to limit the number of channels available over an E1/T1 or PRI ISDN circuit (Fractional T1 line).

The third table allows entry of the MAC address of a remote phone or gateway Module as well as the Nation code and Zone. During registration, the system will compare the MAC address received form the remote device and, if matched, will permit registration of the device. Once the device is registered, the data for the device is placed in the appropriate locations and the data is removed from the Remote Phone & Gateway Registration Table.

Operation

Operation of registration is automatic based on the system database and Registration Switch position.

Conditions

Programming

NUMBERING PLAN	1. MAC Registration Table (PGM 102-Btn 11~14)
TABLES	1. Fractional Module Table (PGM 235)
DEVICE LOGIN	1. Remote Phone & CO Gateway Login, Web Admin only
ZONE	1. Zone Data, (PGM 436~441 &444, Web Admin only)

Related Features

Hardware

2.63 RINGING LINE PREFERENCE

Description

A station is automatically connected to incoming calls by lifting the handset or pressing the **[SPEAKER]** button when assigned Ringing Line Preference.

Operation

iPECS Phone

To answer a call while the station is ringing

1. Lift the handset or press the **[SPEAKER]** button.

Conditions

1. When multiple calls are ringing at the station, a priority defines the order in which calls are answered. The default priority is:
   Transfer > recalls > incoming calls > queued calls
2. Intercom calls are always given the lowest answering priority.
3. Ringing Line Preference overrides Prime Line assignment.
4. SLTs operate only in the RLP mode; when ringing, lifting the handset connects the SLT to the ringing call.

Programming

STATION	1. Auto Speaker Select (PGM111-Btn 1) 2. Ringing Line Preference (PGM 112-Btn 7)
CO/IP	1. Ring Assignments (PGM 144)
SYSTEM	1. RLP Priority (PGM 173)

Related Features

Prime Line Immediately/Delayed
Automatic Speaker Select

Hardware

2.64 SIP EXTENSION SERVICE

Description

The iPECS system supports compatible SIP phones supporting the Internet Engineering Technical Committee standard RFC3261 for real-time communications over the internet. Once registered, iPECS will deliver services to the SIP Phone. Operation of the SIP Phone generally follows the steps outlined for an SLT.

SIP Phones employ a VoIP channel in the system to convert between iPECS and SIP protocol. When configured for Point-to-Point mode, a VoIP channel is not required between two SIP Phones however, the iPECS has no control over the call and features such as Camp-on will not be available.

Operation

Web Admin

To register a SIP Phone:

1. Select Station User Login (PGM 443) in Device Login section.
2. Enter the desired unused station number as the ID using the format “SIP Phone ID=sip:XXXX”, where XXXX is the station number. For Example, the SIP ID=sip:1234 is station number 1234.
3. Click the Save button.

SIP Phone

To set-up the SIP Phone:

1. Configure SIP Phone settings (ex. IP address, Subnet mask, Gateway, Telephone number, Proxy address, Expiration timer, etc.). The Telephone number must be the station number assigned in Station User Login (PGM 443) and the proxy address is the VOIM8/24 IP address.
2. Boot the SIP Phone, which will register it with iPECS.

Conditions

1. The SIP Phone must be in the same IP domain as the serving VOIM8/24 Module. SIP Phone cannot support “Direct Send”.
2. The number of SIP Phones is limited by the station capacity of the system. The number of VOIM channels available limits the number of simultaneous voice/video calls. A SIP Phone uses a VOIM8/24 channel for a voice call. If two (2) SIP Phones establish a voice call, two channels are required.
3. A SIP Phone cannot be configured as a Linked Pair station.
4. A SIP Phone can be a member of conference call, but cannot establish a conference call.
5. A SIP Phone can place a voice over call but cannot receive one.
6. The iPECS Station User name is overwritten by SIP Video Phone’s “Display Name” setting.
7. If the Station number is changed in the iPECS database, the SIP Phone should be reconfigured and re-registered with the system.

8. When SIP Phones are connected in the Point-to-Point mode, the iPECS has no control over the call. Thus, features such as Camp-on are not supported. In addition, Point-to-Point mode is not supported for SIP Phones behind a NAPT.
9. Support for SIP phones requires a license key.

Programming

SYSTEM	1. SIP Station Mode (PGM160-Btn 24)
STATION	1. Station User Login (PGM443 Web only)

Related features

Hardware

VOIM8 or VOIM24
SIP Phones

2.65 SPEED DIAL

2.65.1 Display Security

Description

Station and System Speed Dial numbers may be programmed so that the digits are not displayed on the LCD of iPECS phones.

Operation

To assign Display Security to a Speed Dial number

1. Dial “*” as the first digit of the Speed Dial number.

Conditions

1. The number is displayed when programming the Speed Dial number.
2. Display Security does not affect the SMDR output.
3. Display Security is provided on all CO/IP calls including calls that are transferred or recall.
4. An “*” in any digit position other than the first, will activate Pulse to Tone Switchover.

Programming

Related Features

Dial Pulse to Tone Switchover
Speed Dial

Hardware

2.65.2 Speed Dial Pause Insertion

Description

A pause dialing command may be inserted in a Station or System Speed Dial number. When encountered, the system will stop dialing the Speed Dial number for the assigned "pause" duration. Multiple pauses (**[HOLD]** button depressions) may be inserted into a Speed Dial number.

Operation

System

Pause operation is automatic when encountered; see Station or System Speed Dial for pause entry

Conditions

1. The timed pause is used only with analog CO lines.

Programming

SYSTEM	1. Pause Timer (PGM 181-Btn 10)

Related Features

Station Speed Dial
System Speed Dial

Hardware

2.65.3 Station Speed Dial

Description

Each user can store commonly dialed numbers for easy access using Station Speed Dial bins. With the iPECS-Micro, iPECS-50 and 100, each station has access to 20 Speed Dial numbers and, with other iPECS systems, each station has access to 100 Speed Dial numbers. Each Speed Dial number can be up to 48 digits in length and may include special instruction codes.

Special instruction codes available are:

'*' as 1st digit	Activate Display Security, do not display number.
Flash as 1st digit	Activate dial tone detect.
'#'	Insert a pause dialing command.

'*' not 1st digit	Switch from Pulse to DTMF dialing.
[MSG/CALLBK]	Send as ISDN Keypad Facility IE.
[HOLD]	Insert a Pulse to tone switchover

iPECS Phone users may assign a Flex button for One-Touch access to a specific Speed Dial bin. In addition, the iPECS Phone user may assign a Telephone number directly to a Flex button. In this case, the telephone number is allocated to the highest numbered available Station Speed Dial bin.

Operation

iPECS Phone

*To assign a Flex button as a **{STATION SPEED DIAL}** button*

[PGM] + **{FLEX}** + **[SPEED]** + Station Speed Dial bin number + **[SAVE]**

*To dial using a Station Speed Dial with a **[SPEED]** button*

1. Lift handset or press the **[SPEAKER]** button.
2. Press the **[SPEED]** button.
3. Dial the desired bin number.

To dial using a Station Speed Dial and the LIP 8000 series soft buttons

1. Press the **[DIR]** soft button
2. Press the **[SPEED]** soft button
3. Dial the desired bin number.

*To program a Station Speed Dial number using the **[SPEED]** button*

1. Press the **[PGM]** button.
2. Press the **[SPEED]** button.
3. Dial the Speed Dial bin number.
4. Press the **{CO}**/**{IP}**, **{LOOP}**, or **{POOL}** button or dial the CO/IP line/group access code.
5. Dial the number to be stored.
6. Press the **[SAVE]** button.
7. If desired, enter a name, see alphanumeric entry chart below.
8. Press the **[SAVE]** button.

To program a Station Speed Dial number using the LIP-8000 series soft buttons

1. Press the **[DIR]** soft button
2. Press the **[SPEED]** soft button
3. Press the **[ADD]** soft button
4. Dial the Speed Dial bin number.
5. Press the **{CO}**/**{IP}**, **{LOOP}**, or **{POOL}** button or dial the CO/IP line/group access code.
6. Dial the number to be stored.
7. Press the **[SAVE]** button.
8. If desired, enter a name, see alphanumeric entry chart below.
9. Press the **[SAVE]** button.

SLT

To dial using a Station Speed Dial

1. Lift handset.
2. Dial '558', the SLT Speed Dial access code.
3. Dial the desired bin number.

To program a Station Speed Dial number

1. Dial 561, the SLT Programming code.
2. Dial 558, the SLT Speed Dial access code.
3. Dial the Speed Dial bin number.
4. Dial the CO/IP line/group access code.
5. Dial the number to be stored.
6. Momentarily press the hook-switch.
7. If desired, enter a name, see alphanumeric entry chart below.
8. Momentarily press the hook-switch.

Alphanumeric characters may be entered to name the Speed Dial number using the chart below.

Q – 11 Z – 12 . – 13 1 – 10	A - 21 B - 22 C - 23 2 - 20	D - 31 E - 32 F - 33 3 - 30
G – 41 H - 42 I - 43 4 - 40	J - 51 K - 52 L - 53 5 - 50	M - 61 N - 62 O - 63 6 - 60
P - 71 R - 72 S - 73 Q - 7* 7 - 70	T - 81 U - 82 V - 83 8 - 80	W - 91 X - 92 Y - 93 Z - 9# 9 - 90
Blank - *1 : - *2 , - *3	0-00	#

Conditions

1. Accessing an empty Speed Dial bin will return error tone.
2. A Speed Dial number can use a specific CO/IP Line/Group entered by the user. If the assigned line is busy, a line from the same group will be selected. If all lines in the group are busy, the user may queue for the next available line.
3. All Speed Dial numbers are stored in memory protected from power loss.
4. The CO/IP line used for a Speed Dial is presented to a direct appearance on the iPECS Phone. If there is no direct appearance at the station, a **{POOL}** or **{LOOP}** button is used as the appearance for the line.

5. The user may pre-select the CO/IP line for a Speed Dial number, overriding the CO/IP line assignment in the Speed Dial bin.
6. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
7. If a Speed Dial number contains a “Dial Tone Detect” command, Flash as the first entry in the bin, and dial tone detect is enabled for the CO line, the system must detect dial tone before dialing the Speed Dial number.

Programming

SYSTEM 1. CO Dial-Tone Detect (PGM 160-Btn 3)

Related Features

CO Line Flash
Dial-by-Name
Display Security
Keypad Facility
LNR (Last Number Redial)
Dial Pulse to Tone Switchover
Save Number Redial (SNR)
Speed Dial Pause Insertion
System Speed Dial
Flex Button Direct Speed Dial Assignment

Hardware

2.65.4 System Speed Dial

Description

Commonly dialed numbers can be stored by the System Attendant or by the Administrator in Web Admin for easy access by stations allowed use of System Speed Dial bins. With the iPECS-Micro, iPECS-50 and 100, up to 800 System Speed Dial numbers may be stored. With iPECS 300 models, up to 3000 System Speed Dial numbers are available. With iPECS 600(1200) models, up to 6000(12000) System Speed Dial numbers are available. Each Speed Dial number can be up to 48 characters in length and may include special instruction codes.

Special instruction codes available are:

‘*’ as 1st digit	Activate Display Security.
Flash as 1st digit	Activate dial tone detect.
‘#’	Insert a pause dialing command.
‘*’ not 1st digit	Switch from Pulse to DTMF dialing.

[MSG/CALLBK]	Send as ISDN Keypad Facility IE.
[HOLD]	Insert a Pulse to tone switch-over

iPECS Phone users may assign a Flex button for One-Touch access to a specific System Speed Dial bin.

Operation

iPECS Phone

*To assign a Flex button as a **{SYSTEM SPEED DIAL}** button*

[PGM] + **{FLEX}** + **[SPEED]** + System Speed Dial bin number + **[SAVE]**

*To dial using a System Speed Dial using a **[SPEED]** button*

1. Lift handset or press the **[SPEAKER]** button.
2. Press the **[SPEED]** button.
3. Dial the desired bin number

   iPECS-Micro & 50 & 100: '200'-'999'

   iPECS-300: '2000'-'4999'

   iPECS-600: '2000'-'7999'.

   iPECS-1200: '20000'-'31999'.

To dial a Station Speed Dial number using the LIP-8000 series soft buttons

1. Press the **[DIR]** soft button
2. Press the **[SPEED]** soft button
3. Dial the desired bin number

   iPECS-Micro & 50 & 100: '200'-'999'

   iPECS-300: '2000'-'4999'

   iPECS-600: '2000'-'7999'.

   iPECS-1200: '20000'-'31999'.

SLT

To dial using a System Speed Dial

1. Lift handset.
2. Dial '558', the SLT Speed Dial access code.
3. Dial the desired bin number

   IPECS-Micro & 50 & 100: '200'-'999'

   PECS-300: '2000'-'4999'

   iPECS-600: '2000'-'7999'.

   iPECS-1200: '20000'-'31999'.

Attendant

To program a System Speed Dial number

1. Press the **[PGM]** button.
2. Press the **[SPEED]** button.
3. Dial the Speed Dial bin number

   IPECS-Micro & 50 & 100: '200'-'999'

iPECS-300: ‘2000’-‘4999’
iPECS-600: ‘2000’-‘7999’

4. Press the **{CO}/{IP}**, **{LOOP}**, or **{POOL}** button or dial the CO/IP line/group access code.
5. Dial the number to be stored.
6. Press the **[SAVE]** button.
7. If desired, enter a name, see alphanumeric entry chart under Station Speed Dial.
8. Press the **[SAVE]** button.

To dial a Station Speed Dial number with the LIP-8000 series soft buttons

1. Press the **[DIR]** soft button
2. Press the **[SPEED]** soft button
3. Press the **[ADD]** soft button
4. Dial the Speed Dial bin number
   IPECS-Micro & 50 & 100: ‘200’-‘999’
   iPECS-300: ‘2000’-‘4999’
   iPECS-600: ‘2000’-‘7999’
   iPECS-1200: ‘20000’-‘31999’.

5. Press the **{CO}/{IP}**, **{LOOP}**, or **{POOL}** button or dial the CO/IP line/group access code.
6. Dial the number to be stored.
7. Press the **[SAVE]** button.
8. If desired, enter a name, see alphanumeric entry chart under Station Speed Dial.
9. Press the **[SAVE]** button.

Conditions

1. Accessing an empty Speed Dial bin will return error tone.
2. A Speed Dial number can be assigned to use a CO/IP Line/Group. If the assigned line is busy, a line from the same group will be selected. If all lines in the group are busy, the user may queue for the next available line.
3. All Speed Dial numbers are stored in memory protected from power loss.
4. The CO/IP line used for a Speed Dial is presented to a direct appearance on the iPECS Phone. If there is no direct appearance at the station, a **{POOL}** or **{LOOP}** button is used as the appearance for the line.
5. The user may pre-select the CO/IP line for a Speed Dial number, overriding the CO/IP line assignment in the Speed Dial bin.
6. A name can be entered for a Speed Dial number to permit access from the Dial-by-Name directory.
7. If a Speed Dial number contains a “Dial Tone Detect” command, Flash as the first entry in the bin, and dial tone detect is enabled for the CO line, the system must detect dial tone before dialing the Speed Dial number.

Programming

STATION 1. Speed Dial Access (PGM 112-Btn 8)

SYSTEM 1. CO Dial-Tone Detect (PGM 160-Btn 3)

TABLES 1. System Speed Zone (PGM 232)

Related Features

CO Line Flash
Dial-by-Name
Display Security
Keypad Facility
LNR (Last Number Redial)
Dial Pulse to Tone Switchover
Save Number Redial (SNR)
Speed Dial Pause Insertion
System Speed Dial
System Speed Zone (Groups)

Hardware

2.65.5 System Speed Zone (Groups)

Description

System Speed Dial bins can be divided in to zones or groups. iPECS systems have 10 zones available, except for the iPECS 600(1200), which has twenty (fifty) zones. To access a Speed Dial bin assigned to a zone, the station must be allowed access to the Speed Dial Zone.

In addition, each zone can be assigned to apply COS dialing restrictions to the Speed Dial numbers in the zone. When assigned, the system will apply the Station and CO/IP COS to calls using Speed Dial numbers in a COS enabled zone.

Operation

Refer to System Speed Dial for operation.

Conditions

1. The conditions of System Speed Dial apply to System Speed Dial numbers assigned to a zone.
2. A station not assigned access to a zone will receive error tone when trying to access a System Speed Dial number in the zone.
3. The Station COS is applied to the Speed Dial number based on the active service mode, Day/Timed or Night.

Programming

STATION 1. Speed Dial Access (PGM 112-Btn 8)

TABLES 1. System Speed Zone (PGM 232)

Related Features

System Speed Dial
Class of Service

Hardware

2.66 STATION CALL COVERAGE

Description

The Call Coverage feature permits an iPECS Phone user to receive ring and answer calls directed to a covered station. This feature is generally employed to allow a Secretarial answering position to cover calls to other stations. When a covered station rings, the **{CALL COVERAGE}** button LED will flash and the covering station may receive ring (immediate or delayed) for the call. The covering station can answer the call using the **{CALL COVERAGE}** button, terminating ring at other stations. Once answered, the LED of **{CALL COVERAGE}** buttons for the station at other covering stations will extinguish.

Operation of this feature requires a **{CALL COVERAGE}** button at the covering iPECS Phone and the covered station must activate call coverage. A station can have multiple Call Coverage buttons each covering a different station and multiple stations can have a Call Coverage button for a given station.

Operation

iPECS Phone

*To assign a **{CALL COVERAGE}** button at the covering station*

[PGM] + **{FLEX}** + '*67' + covered Station number + **[SAVE]**

To activate Call Coverage at the covered station

4. Press the **[PGM]** button
5. Dial '141', the Call Coverage code.
6. Dial '1' to enable or '0' to disable Call Coverage.

*To assign ring for a **{CALL COVERAGE}** button*

1. Press the **[PGM]** button
2. Dial '142', the Call Coverage Ring code.
3. Dial the delay in ring cycles ('0'-'9').

When a covered station receives a call, the covering station will receive the following display:

CALL FOR STA xxxx FROM yyyy time

Condtional Operation by Administrative Attributes

1. *'Call Coverage On Busy' =*
   OFF : Call Coverage is implemented when a covered station is on idle state
   ON : Call Coverage is implemented when a covered station is on idle or busy state
2. 'Call Coverage Through Mobile Ext' =
   OFF : Call Coverage Ring is not delivered to Mobile Extension
   ON : Call Coverage Ring is not delivered to Mobile Extension
3. 'Call Coverage On Busy Range' =
   External Call Only : Call Coverage only for external call when covered station is busy.
   External and Internal Call : Call Coverage both for external & internal call when covered station is busy.
4. 'Call Coverage Delay Ring Method' =
   by Originator : Call Coverage Delay is implemented by covered station' 'Call Cover Delay Ring' delay count'
   by Member : Call Coverage Delay is implemented by covering station' 'Call Cover Delay Ring' delay count'
5. 'Call Coverage For Wakeup Ring' =
   OFF : Wakeup Ring to covered station is not covered
   ON : Wakeup Ring to covered station is covered

Conditions

1. An iPECS Phone user may cover for an SLT or other stations. However, since a Flex button is required, an SLT cannot provide coverage for other stations.
2. When off-hook or in DND, the covering station will only receive a visual indication of the call from the LED of the **{CALL COVERAGE}** button and display, no off-hook ring is provided.
3. The **{CALL COVERAGE}** button will provide an appearance for CO/IP lines that do not appear on the covering station except for Private Lines. To cover for Private Lines, the covering station must have an appearance and be allowed access to the Private Line.
4. Call Coverage attributes can be assigned either by the covered station user or in the system database.
5. Link-Paired Station does not receive call coverage ring on both the original and paired station, since they have their own flexible button and call coverage feature is not implemented on same station but different flexible buttons for same covered station.

Programming

STATION

1. Call Coverage Enable (PGM 111-Btn 12)
2. Call Coverage Ring Delay Count (PGM 111-Btn 13)
3. Call Coverage On Busy (PGM 111, OFF/ON)
4. Call Coverage Through Mobile Ext (PGM 111, OFF/ON)

5. Call Coverage On Busy Range (PGM 111, External Call Only/ External and Internal Call)
6. Call Coverage Delay Ring Method (PGM 111, by Originator/by Member)
7. Call Coverage For Wakeup Ring (PGM 111, OFF/ON)

Related Features

Hardware

2.67 STATION GROUPS

Description

Stations can be grouped for incoming call routing and Call Pick-up purposes. Ten types of groups can be defined:

Circular
Terminal
ACD
Ring
Pick-Up
External Voice Mail
VMIM/VSF-Voice Mail
Feature Server UMS Group
Net VM (Centralized External VM)
Unified Communication Solution Server

Circular Station Group

In Circular Hunt, calls to a station in the group will go to the station, if unavailable or unanswered in the hunt no answer time; the call will be directed to the next station defined in the group. The call will continue to hunt until each station in the group has been tried. The call remains at the last station or passes to a designated overflow station or group.

A Circular Station Group can be assigned with a pilot number (the Station Group Number) so that calls to the pilot number will hunt. In this case, the call will be directed to the first station in the group and, if needed, hunt through each station in the group until reaching the last station. The call may remain at the last station, passed to an overflow destination or sent to a voice mailbox.

Terminal Station Group

Calls to a station in a Terminal Station Group that encounter an unavailable or unanswered status will be routed through the hunt process. The call will proceed to the next listed station in the group until reaching the last listed station in the group. The call may remain at the last station or be routed to an Overflow destination.

A Terminal Hunt Group can be assigned with a pilot number (the Station Group number) so that calls to the pilot number will hunt. In this case, the call will route as described for Circular Pilot Number hunting.

ACD Station Group

Calls can be sent to an ACD group by dialing the Station Group Number or assigning CO/IP lines to ring directly to the Station Group. Calls are directed to the station in the group that has been idle for the longest continuous time, Uniform Call Distribution. If all stations are busy or unavailable when the call is received, the call may be routed to an alternate location or may continue to wait (queue) for an available station in the group. After queuing to the group, the caller may be routed to an overflow destination, which can be a Station, Station Group or Voice Mailbox.

An ACD supervisor can be assigned to monitor the group and act to oversee operations of the group. The ACD Supervisor can print group statistics and activate alternate routing as well as assist agents.

Station Ring Group

A call to any station in the Group will cause all stations in the group to ring and any station in the group may answer the call. If the call remains unanswered beyond the Overflow timer, the call is sent to the Overflow destination, which can be a Station, Station Group or Voice Mailbox.

Multiple calls can be received by a Station Ring Group and can be serviced in any order.

External AA/VM Station Group

This group is assigned to support an external Auto Attendant Voice Mail system that employs SLT ports to interface to the iPECS. An External AA/VM group is assigned for either circular or terminal hunt. The External AA/VM may employ either in-band signaling over the audio channel or SMDI protocol with a signaling connection to the system RS-232 channel.

Pick-Up Station Group

A station can be assigned to a Call Pick-Up group and then may pick-up (answer) calls to other stations in the group employing the system's Group Call Pick-Up feature.

VMIM/VSF AA/VM Group

The VMIM/VSF memory is employed by the integrated iPECS AA/VM application. Incoming calls can be directed to one of 70 user-recorded announcements, which may request further routing instructions from the user in the form of caller dialed digits. These digits are employed to route the caller as defined in the system CCR (Customer Controlled Routing) Tables.

The VMIM/VSF AA/VM Group Voice Mail application receives calls forwarded or recalling from a station. Such calls will receive the user's pre-recorded greeting and may leave voice messages. The user may call the VMIM/VSF AA/VM Group to review and manage the integrated Voice Mail application.

Feature Server UMS Group

The Feature Server is a PC based TAPI application with high-end Auto Attendant, Voice Mail and Unified Messaging Service (Voice/Fax and e-mail). The iPECS Feature server receives calls, plays announcements, stores voice messages and forwards them as wave file attachments to the user's e-mail. The application also receives Faxes and forwards them as attachments to e-mail. The Text-to-Speech option permits listening to e-mails as well as voice mails.

Net VM

This group is defined to support a Centralized Voice Mail system for a networked environment. At supported systems, the group is used to handle the AA/VM requirements from the central iPECS. The Net VM group may be an external VM system or the iPECS Feature Server.

UCS Group

This group is defined to support the Unified Communication Solution available with the iPECS system.

Group Announcements

Station Group routing can be augmented with announcements recorded in the VMIM/VSF AA/VM. Callers can be routed to one of several user-recorded announcements. The system answers the call and plays the defined 1st announcement to the caller. The announcement may provide the caller with routing options for Caller Controlled Routing. The 1st announcement may be "Guaranteed" meaning it will play in full before routing the call. A 2nd announcement can be provided to the caller should queue timers expire.

Operation

Conditions

1. The number of groups and member stations per group are given in Table 1.2-1.
2. Station Group calls are not routed to member stations that are in DND.
3. When a member of a Circular, Terminal, or Ring Group activates Call Forward, calls to the group may still route to the member based on the Member Forward option.
4. A call transferred to a Station Group will follow the routing for the group and will not initiate the Transfer Recall process.
5. Calls to a Station Group receive either ring-back tone or MOH while queued to the group.
6. Calls, which are not answered in the Overflow time, are routed to the defined Overflow destination, station, group, etc. If no Overflow destination is defined, the call is dropped after expiration of the Overflow timer.

7. One of the 70 VMIM/VSF announcements may be assigned as the Overflow destination. These announcements allow for Caller Controlled Routing.

Programming

STATION GROUPS
1. Station Group Assignment (PGM 190)
2. Station Group Attributes (PGM 191)
3 Pickup Group (PGM 192)

Related Features

Automatic Call Distribution
External Auto Attendant/Voice Mail
Group Call Pick-Up
MOH (Music-On-Hold)
VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

2.68 RING GROUPS INDICATION

Description

If a station call to Ring group and a member of the Ring group received the call, the member can hear mute ring and see the flex button flashing which is the station, caller. And after that the member of Ring group can push the button and talk with the caller and talking internal call can be hold. After completing the talk, you can seize again holding station.

Operation

iPECS Phone

*To assign a **{STATION NUMBER}** button of caller station at the called station*

[PGM] + **{FLEX}** + calling Station number + **[SAVE]**

To hold and answer a calling station

1. A user is Talking with another station
2. The other Caller call to Ring group.
3. The member of ring group can hear the mute ring and flashing the button of caller.
4. The user can press the button and talk with the caller who call to Ring group.
5. And the call of previous talking is holding.
6. The user can press ICM button or flex button for reconnecting with previous call.

Conditions

1. This operation should be operated only by programming.

Programming

SYSTEM 1. Ring-Group Indication (PGM 161-B24-B16)

Related Features

Ring Group Call Pick-Upservice ON

Hardware

2.69 SMDR (Station Message Detail Recording)

2.69.1 Call Cost Display

Description

Each SMDR call record includes a "Cost" field, which is a calculated estimate for the cost of the call. The call cost updates in real-time and displays on the iPECS Phone LCD in place of the call duration.

The cost is determined by:

1. Fixed charge per "Call Meter Pulse",
2. ISDN Advice of Charge or
3. Estimated cost updated based on Elapsed Call Timer and assigned costing.

The technique selected to determine cost is based on the type of facility (analog CO, ISDN, or VoIP), services provided by the carrier and the system database.

<u>Analog CO</u>

Where "Call Metering Pulse" service is available from the carrier, the system will apply the "SMDR Cost per Unit Pulse" and the "SMDR Decimal" to the Call Metering received to estimate the call cost.

When no "Metering Type" is selected, the system call duration is used with the cost/pulse and decimal values to estimate the cost of the call. The cost is updated periodically using the "Elapsed Call Timer" duration.

<u>ISDN</u>

ISDN providers may support "Advice of Charge" information in the ISDN Facility Message. If assigned, the system will employ this information to display and output the call cost.

<u>VoIP</u>

For VoIP calls, the system employs the call duration, cost/pulse and decimal values to establish the call cost estimate. The cost is updated periodically at intervals of the "Elapsed Call Timer".

Operation

System

Call cost is estimated automatically and output to iPECS Phone displays and the SMDR RS-232 port

Conditions

1. The call cost display begins after the "SMDR Start Timer" expires, if enabled, or at receipt of the first Call Meter Pulse.
2. Once connected to the system, the call duration includes the total time the call is connected including periods when the call is on hold, in queue, etc.
3. To enable Call Cost Display, the "SMDR Cost per Unit Pulse" and "SMDR Decimal" must be assigned. When not assigned, the call duration is provided by the system.

Programming

CO/IP	1. Metering Type (PGM 142-Btn 3)
SYSTEM	1. SMDR Cost per Unit Pulse (PGM 177-Btn 10) 2. SMDR Decimal (PGM 177-Btn 11) 3. SMDR Start Timer (PGM 177-Btn 12) 4. CO Warning Tone Timer (PGM 180-Btn 19)

Related Features

SMDR (Station Message Detail Recording)
Lost Call Recording
Traffic Analysis
CO/IP Call Warning Tone Timer

Hardware

RS-323 device to capture SMDR

2.69.2 SMDR Call Records

Description

SMDR (Station Message Detail Recording) provides detailed information on incoming and outgoing calls. Assignable options in the system database permit recording of all external calls, all external outgoing calls or toll calls, calls that exceed a fixed duration and intercom calls. Call records are output either upon completion of the call (real-time) or in response to a request from the System Attendant. SMDR may be sent periodically via e-mail to a defined e-mail address.

The SMDR record output for external calls is as shown in the figure below. There are two flexible fields, Field I and Field II. Each Field is defined to show Ring duration, CLI (Caller Id) or CPN (Called Party Number).

```
NO   STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II DC
XXXX SSSS BBB DD:DD FF/FF/FF EE:EE HCCCCCCCCCCCCCCCCCC sssss aaaaaaaaaaaa hcccccccccccccccccc rr
```

The various fields or items for a Call Record are:

NO SMDR record number, optional output, (PGM 177-Btn 23)
STA 3 or 4 digit station number.
CO 2 or 3 digit CO Line number
Time Call duration in minutes and seconds
Start Date and time call was placed/received
NUM I Flex Field I, for outgoing call displays dialed number, for incoming call displays Ring duration, CLI or CPN (PGM 177-Btn 20
Cost Cost of Call
Account Code Account code entered for call
NUM II Flex Field II for incoming call, displays Ring duration, CLI or CPN or blank, (PGM 177-Btn 23)
DC Disconnect Cause, optional output (PGM 177-Btn 29), with value:
01~7F: Follow cause value of Q.931 specification,
81: Disconnected, toll restricted
84/85: Disconnected, disconnect supervision
0: Other

The SMDR record output for internal calls is as shown in the figure below.

```
---- Site Name :              ----------------------------------------------------------------------
 STA  CO   TIME         START        DIALED   NUMBER          COST      ACCOUNT CODE        CPN   NUMBER
----- --- -------- -------------- --------------------- ---------- ----------- --------------------
102       00:00:01 06/11/08 10:53 E103
102       00:00:00 06/11/08 14:13 S103
102       00:00:10 06/11/08 11:05 E Conf Rm 1
120       00:00:03 06/11/08 11:11 P123
```

STA	DIALED	Meaning
100	E101	1) Station 100 called station 101 and station 101 answered. 2) In a conference, station 100 was a conference master, station 101 was a conference member.
105	S103	Station 105 sent an internal SMS to station 103
110	E Conf Rm 1	Station 110 was the member of conference room 1.
120	P123	Station 120 paged station 123

Operation

System

For real-time SMDR, records are output after completion of the call as shown in the figure above

System Attendant

To print SMDR records

1. Press the **[PGM]** button.
2. Dial '0111', the SMDR print code.
3. Enter the desired station range.
4. Press the **[SAVE]** button.

To delete stored records

1. Press the **[PGM]** button.
2. Dial '0112', the SMDR delete code.
3. Enter the desired station range.
4. Press the **[SAVE]** button.

To abort SMDR printing

1. Press the **[PGM]** button.
2. Dial '0114', the SMDR abort code.
3. Press the **[SAVE]** button.

Conditions

1. For SMDR, if the first dialed digit(s) match the programmed LD code or the number of dialed digits exceeds the LD digit count, the call is considered an LD call. When behind a PBX, LD determination is made only if a PBX Trunk Access code is dialed as the first digit(s).
2. Except for DISA calls, the duration of ring for an incoming call is provided in the Dialed number field.
3. A header, including the assigned "Customer Site Id" is output after two blank lines and is repeated every 66^{th} line.
4. The SMDR output is a simple ASCII stream of up to 80 characters per line.
5. When enabled, SMDR call record timing begins after the "SMDR Start Timer" expires and ends at call completion.
6. For security, if an Authorization code is entered as the Account code, the call record will show "STA-P" and the station number, or "SYS-P" and the Authorization code bin number in place of the Account code.
7. For incoming calls, the "NUM I" and "NUM II" fields will display the assigned data item – Ring Service time, CLI, or CPN. For outgoing calls, the NUM I field will always display the dialed number, user or system.

Programming

SYSTEM	1. SMDR Attributes (PGM 177)
	2. Field I (PGM 177-Btn 20)

3. Field II (PGM 177-Btn 22)
4. Print Serial No (PGM 177-Btn 23)
5. SMDR DISC Cause (PGM 177-Btn 24-5)
6. SMDR ICM SAVE (PGM177-Btn 24-3)
7. SMDR ICM PRINT (PGM177-Btn 24-4)

Related Features

Call Cost Display
Lost Call Recording
Traffic Analysis
Authorization Codes (Password)

Hardware

RS-323 device to capture SMDR

2.69.3 Lost Call Recording

Description

Incoming calls where the caller hangs up before answer or while in a hold state are considered Abandoned or Lost calls. Special SMDR call records are provided for lost calls in real-time, as they occur, and a summary Lost Call count report is available on demand.

The real-time Lost Call records provide details on the called party, when and how long the call rang or was on hold before being abandoned, etc. Description of the record details is provided in the following charts. As noted in the charts, the dialed number field indicates the type of call and the ring or hold duration before the call was abandoned. The first character in the NUM I field is the status of the call when abandoned:

R normal ring to a station,
G ring to a station group and
H call placed in a hold state, including Transfer hold.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
EXT  31  00:00 14/05/02 15:45 R RING 01:35
```

➢ Incoming call on CO Line 31 received on May 14, 2002 at 3:45 pm, rang the assigned stations for 1 minute and 35 seconds.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
101  31  00:00 14/05/02 16:45 R RING 02:03
```

- Station 101 rang for an incoming call on CO Line 31 on May 14, 2002 at 4:45 pm, rang for 2 minutes and 3 seconds.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
101  02  00:00 15/05/02 09:35 R 100 RING 00:49
```

- Incoming call on CO Line 02 on May 15, 2002 at 9:35 am forward from station 101 to station 100 and rang for 49 seconds.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
104  05  00:00 16/05/02 11:06 G621 RING 01:32
```

- Incoming call on CO Line 05 on May 16, 2002 at 11:06 am routed to station 104 of Station Group 620 and rang for 1 minute and 49 seconds.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
621  17  00:00  16/05/02 14:03 G621 RING 00:39
```

- Incoming call on CO Line 17 on May 16, 2002 at 2:03 pm routed to Station Group 621 and rang for 39 seconds.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
100  01  03:32 16/05/02 15:30 H100 03:02
```

- Call on CO Line 1 on May 16, 2002 at 3:30 pm placed on hold by station 100 for 3 minutes and 2 seconds had total duration of 3 minutes and 32 seconds.

```
STA  CO  TIME  START          DIAL/CLI/CPN NUM-1  COST  ACCOUNT CODE DIAL/CLI/CPN NUM II
129  23  00:45 18/05/02 08:40 H100 RING 00:33
```

- Call on CO Line 23 on May, 18, 2002 at 8:40 am was transferred by station 100 to station 129 was on hold for 33 seconds.

The output for the Lost Call summary count report is shown in the figure below:

```
Lost call count start time: 05/01/02 09:31
Current time 26/04/02 16:32
Total Lost call count until now: 121
```

Operation

System Attendant

To print the summary Lost Call Count report

1. Press the **[PGM]** button.
2. Dial '0115', the Lost Call Count report code.
3. Press the **[SAVE]** button.

To reset the Lost Call summary Count

1. Press the **[PGM]** button.
2. Dial '0116', the Lost Call Count Reset code.
3. Press the **[SAVE]** button.

Conditions

1. When the Lost Call Count is reset, the SMDR port will provide a "count reset" message.
2. Individual Lost Call records are only available in real-time and not on-demand.
3. "Print Incoming Calls" and "Print Lost Calls" must be enabled in the SMDR Attributes for the system to output real-time Lost Call records and for the Lost Call Count summary report.
4. The fields of a Lost Call Record are the same as a normal SMDR Call Record.

Programming

SYSTEM 1 SMDR Attributes (PGM 177)

Related Features

Call Cost Display
SMDR Call Records
Traffic Analysis

Hardware

RS-323 device to capture SMDR

2.70 SYSTEM ADMIN PROGRAMMING

2.70.1 Keyset Administration

Description

The system database can be accessed and modified with the keypad and Flex buttons of an iPECS Phone. The display of the iPECS Phone is employed to view items in the iPECS database. The user may be required to enter a password for access to Keyset Admin. Based on the Multi-level password, the user has access to specified system database program codes.

For detailed information on database administration and maintenance, refer to the **iPECS Admin & Program Manual**.

Operation

Operation is detailed in the iPECS Admin and Maintenance Manual.

Conditions

1. Only stations assigned with Admin access can enter and change the system database items. As a default, all iPECS phones can access the database.

Programming

STATION
1. Admin Access (PGM 113-Btn 1)

SYSTEM
1. Admin Access Authority (Web only)
2. System Password (PGM 162)

Related Features

Web Administration
Multi-Level Admin Access

Hardware

2.70.2 Multi-Level Admin Access

Description

Access to the Admin database is password protected. Up to three (3) levels of access to the database can be established by assigning a different password to each level. The Maintenance level has access to the entire database, all maintenance routines and defines the Admin Access Authority of the two remaining passwords. The User password can only access the defined database items and cannot access the Station Program pages. The Admin password has access to the defined database items as well as Station Program pages. In Web Admin, the Maintenance password user establishes the Access Authority for each password selecting the Admin Program Codes available each password level.

Operation

Detailed operation of Admin access and assigning access authority for each level is given in the iPECS Admin & Maintenance Manual.

Conditions

1. Admin Access Authority is defined only in the Web Admin; it cannot be defined when using Keyset Admin
2. Admin Access Authority applies to all Admin access whether accessed via an iPECS Phone, ISDN or IP channels.

Programming

STATION
1. Admin Access (PGM 113-Btn 1)

SYSTEM
1. Admin Access Authority (Web only)
2. System Password (PGM 162)

Related Features

Keyset Administration
Web Administration

Hardware

2.70.3 Web Administration

Description

The system database is accessed and modified via an iPECS Phone, the LAN interface or an ISDN BRI/PRI line. Both the LAN and ISDN access the system's Web server, which delivers the database as a set of Web pages to the user's Web browser. Under the proper conditions, both also allow for remote access to the system database.

For remote access with the LAN interface, the system must be assigned a remotely accessible IP address. The IP address should be fixed either as a public IP address or through a NAPT server with port forwarding. The browser should be pointed to the system's IP address and the TCP port assigned in the system database.

For the ISDN BRI/PRI, a PPP (Point-to-Point Protocol) connection can be established between a standard ISDN modem and the iPECS BRI/PRI module. The connection can be set-up over a normal or DID BRI/PRI line. The system negotiates access using 2 User Ids and matching passwords assigned in the database. In addition, for DID access the Called Party number must match the assigned system PPP destination. Once a connection is established over the BRI/PRI, the user's Web browser can be opened and pointed to the system's IP address and assigned port for access to the database.

When accessed, the system will deliver the iPECS Admin and Maintenance Web page. From this page, selecting Admin & Maintenance will return the login page where the user must enter a password that matches an assigned password. Based on the password entered, the user is permitted access to specified system program codes.

For detailed information on database administration and maintenance, refer to the **iPECS Admin & Program Manual**.

Operation

Operation is detailed in the iPECS Admin and Maintenance Manual.

Conditions

1. For Web Admin, the password can be encrypted using the LG-Ericsson Java Virtual Encryption plug-in. A Java Virtual Machine (MS or Sun) must be installed in the user's PC to support password encryption.

Programming

NUMBERING PLAN	1. System IP Address Plan (PGM Code 102)
SYSTEM	1. Web Server TCP Port (PGM 161-Btn 14) 2. WEB Password Encryption (PGM 161-Btn 15) 3. Admin Access Authority (Web only)
ISDN/ICLID	1. BRI/PRI PPP Web Admin Attributes (PGM 205)

Related Features

Keyset Administration

Multi-Level Admin Access

Hardware

2.70.4 Web User Manual

Description

The iPECS User Guide is available on-line as part of the iPECS Web services. The iPECS main Web page permits access to the Web User guide. The guide is an html document, which can be modified by replacing the HTML ROM image in the system with an external ROM image in accordance with the appropriate LG-Ericsson R&D STI.

Operation

Operation is detailed in the iPECS Admin and Maintenance Manual.

Conditions

1. Access to the User Guide is not password protected.
2. To support local languages, the HTML ROM image of a translated Web User guide may be loaded into system memory.

Programming

Related Features

Web Administration

Hardware

2.71 SYSTEM NETWORKING

2.71.1 Centralized Control TNET

Description

In a Centralized Control TNET (Transparent Network), a central MFIM controls all remote modules and terminals providing transparent networked access to all the features and functions of the central iPECS as well as the resources connected to the iPECS.

Where the remote device is not directly reachable by the iPECS, RTP packets must be relayed through a local VoIP channel. A remote device may not be reachable when WAN access for the device is through a firewall or NAPT server. In this case, the remote devices are assigned a zone to manage RTP traffic between other devices connected in the TNET. The zone defines when an individual device requires use of the local VoIP channel. Zones are used to identify other group characteristics as outlined in section Remote Device Zone Management.

Remote sites may include an MFIM operating in the local mode as a live back up to the remote central system. Under normal circumstances, the central MFIM controls remote devices (gateway Modules and terminals) including any local MFIM VoIP channels. However, should the WAN connection between the central system and the remote devices fail, the local MFIM will assume the call server responsibility for the local devices. The local MFIM thus provides local survivability and, based on configuration, may provide PSTN back-up service (Fail-over) for internal calls that normally route over the WAN.
Under certain operating conditions, this equipment cannot be relied upon to make emergency calls. Alternative arrangements should be made for access to the emergency services.

Operation

System

Operation of Centralized Network is automatic when configured & defined

Conditions

1. In a Centralized Network, the maximum number of channels available is the maximum number of channels supported by the central MFIM.
2. In TNET, Centralized Miscellaneous functions (Relay support, MOH, BGM, Alarms and External Page) are not supported but, all terminals in the TNET can make and receive pages.
3. When NAPT or other firewall functions are implemented, packet relay for RTP packets is required. Packet relay requires VoIP channels for each simultaneous call desired.
4. The local MFIM will take over operation of registered devices if the central controlling MFIM does not respond to three consecutive poll attempts over a 10-second period. The central MFIM will gain control automatically upon return of the WAN connection.
5. iPECS can be installed behind a NAPT however, Fixed Nat – port forwarding is required for the host to be reachable by remote devices.
6. Up to 50 local MFIMs can operate in local mode as part of a TNET if cetral MFIM is MFIM1200.
   Otherwise, Up to 15 local MFIMs can operate in local mode as part of a TNET.
   Up to 30 VSF/VMIM could be used in MFIM1200.

7. You can set TNET enable ON/OFF for using TNET in local MFIM

Programming

TNET

1. TNET Basic Attributes (PGM 330)
2. TNET Central MFIM Attributes (PGM 331)
3. TNET Local MFIM (PGM 332)
4. Fail-over PSTN Attributes

Related Features

Remote Device Zone Management

Hardware

Remote MFIM to support Local Survivability

2.71.2 Distributed Control Network

Description

In the Distributed Control Network, each iPECS system maintains control over the devices registered to it. The networked systems communicate allowing other networked systems access to resources over the network. In addition, other features and functions as detailed in the following sections of this

manual are available to users in a distributed network environment. The iPECS permits remote access to various resources through registered gateway Modules and terminals.

In addition, iPECS will request access to resources of remote systems. The user-dialed number is analyzed and the call routed according to the Net numbering table. Should the main path fail to respond, the iPECS routes the call employing the alternative Speed Dial route assigned.

iPECS supports two (2) standard protocols, QSIG over ISDN and H.450 over IP, for the basic networking functions and the proprietary iPECS protocol for the advanced networking features. QSIG employs ISDN PRI channels only with support for ESTI standards ETS 300-237/238/256/257/260/261/361/362/363/364.

Operation

Operation of Distributed Networking is automatic when configured & defined

Conditions

1. To use the networking features, the software lock-key installation is required. Two types of software lock-key relate to the networking protocols. One is for QSIG based networking protocol, and another is for VOIP based networking protocol. Each iPECS system has a unique software lock-key. To get the software lock-key, contact the distributor of iPECS system.
2. Unified Dialing Plan (UDP): Each station can have a unique number up to 7 digits in the networked systems, but it depends on their own numbering plan.
3. The alternative route employs a Speed Dial number to place a call and is not a Networked call. Thus, the Distributed Control Network features are not available.

Programming

Related Features

Hardware

2.71.2.1 Net Call

Description

A station user can make a call to a station in other systems by dialing only a station number just as an intercom call within the same system.

Operation

1. Lift Handset or press the **[SPEAKER]** button. The system provides a user with a dial tone.
2. Dial the station number of other systems, or press the {NET DSS} button of other systems.
3. The station seizes the network CO line according to the net routing table, and the system sends a digit stream that is modified by the net routing table.
4. The called party receives a digit stream that is sent by calling party, and analyzes it using the net routing table to determine
5. The right destination. The called station receives a ringing signal.
6. The LED of [Network CO] button will be extinguished when the Net Call is cleared.

Conditions

1. Net call must be used without seizing a CO line.
2. User hears an error tone if there is no idle networking path.
3. In spite of ICM mode, the called party receives a ringing signal for the networking call.
4. When system detects the fatal error from the network, system sends the digit stream to the network using the alternate speed dial bin. In this case, the call is not a networking call.
5. The Net Call is also applied CO Call Restriction Timer (PGM180-Btn14).

Programming

SYSTEM

1. Networking Basic Attributes (PGM 320)
2. Networking CO Line Attribute (PGM 322)
3. Network Numbering Plan Table (PGM 324)

Related Features

Hardware

2.71.2.2 Net Transfer

Description

A station user can transfer any kind of CO line to a station in other systems by pressing **[TRANS]** button and dialing a transferred station such as a call transfer within the same system. There are two kinds of transfer, screened and unscreened transfer.

Engineering notice) There are two kinds of standard transfer method in QSIG and H.450; Transfer by join and Transfer by rerouting. The main difference is how control the connecting path between transferring and transferred station. In case of Transfer by join, additional connecting path will be needed to transfer the call to another station. In case of Transfer by rerouting, new connecting path is used to transfer the call and old connecting path of transferring station will be cleared.

Operation

Screened transfer

1. Press the **[TRANS]** button at a station during conversation with a CO line. The CO line is placed on Exclusive Hold.
2. Dial the station number of another system to transfer the call. The transferred station of another system receives a ring signal.
3. Announce when the transferred station answers. Both stations can make a conversation each other, but the held CO is still in waiting on Transfer hold.
4. Hang-up to complete the transfer.

Unscreened transfer

1. Press the **[TRANS]** button at a station during conversation with a CO line. The CO line is placed on Exclusive Hold.
2. Dial the station number of another system to transfer the call.
3. Hang-up to complete the transfer.

Conditions

1. If both of transferred and transferred-to stations are located in the same system, the networking path will be cleared. That is, the transfer call will be setup as intercom call.
2. The transfer will be canceled when user presses the flashing [CO] or **[TRANS]** button.
3. Net Transfer call does not recall to the origination.
4. User hears an error tone if there is no idle networking path.
5. Net transfer is not activated to a busy station.

Programming

SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Networking Supplementary Attribute (PGM 321 – BTN 1)
3) Networking CO Line Attribute (PGM 322)
4) Network Numbering Plan Table (PGM 324)

2.71.2.3 Identification Service

Description

Calling Name Identification Presentation (CNIP): When a user makes a net call and a name of station is programmed in the Station Name field (PGM+74), the system includes the name of calling party to the called party between systems.

Operation

1. A Net Call is arrived a station with LCD display.
   While ringing, the CNI will be displayed if they are included in the Setup message.

Conditions

Programming

SYSTEM Network CNIP/COLP Enable (PGM 320 – BTN 3 / BTN 4)

2.71.2.4 Call Completion

Description

There are two kinds of call completion as follows;

Completion of Calls to Busy Subscribers (CCBS):
After calling a user in another system using basic call and encountering a busy tone. A station user can be notified when the busy destination of another system becomes idle. If the user wants to make a call to the destination on that notification, the call can be reinitiated to the destination of another system again.

Completion of Calls on No Reply (CCNR):
After calling a user in another system using basic call and encountering no reply. The caller can be notified when the destination becomes an idle status after some actions. If the caller wants to make a call to the destination, the call can be reinitiated to the destination again.

Operation

To make CCBS (Call Back)

1. Dial the station of another system that is a busy.
2. Press the [CALLBK] button while a busy tone is provided.
   The call is cleared after a confirmation tone.
3. The busy station goes to Idle; the originator receives a call-back ring.
4. When the originator answers to the call-back ring, a new call will be activated to the calling station.

Conditions

1. Stand-alone IP Phone that supports H.450 can activate the Call Completion feature.

2. A station can leave or have only one callback message, and a new request will be left message wait indication message on busy station.
3. A voice message cannot be left even though the VMIM/VSF is installed in a local system.
4. When the originator does not answer the call back ring within net timer, the call will be cleared.
5. There are two modes: One is connection mode and the other is connectionless mode. This can be selectable at PGM 320/BTN 8.

Programming

SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Networking CO Line Attribute (PGM 322)
3) Network Numbering Plan Table (PGM 324)

2.71.2.5 Call Offer

Description

A busy user on one node is given notification that another call is waiting from another node. It is similar to a Camp-On function.

Operation

To activate Call Offer

1. Dial a busy station number of another system. The caller hears a busy tone.
2. Press the [CAMP ON] button or '*' during hearing a busy tone.
   The busy station receives an off-hook muted ring.
   The calling station hears a ring-back tone instead of a busy-tone.

To answer the Call Offer

1. Press the flashing CO line button while receiving a muted ring.

Or,

2. The muted ring is changed to normal CO ring when you go on-hook state. Then you can answer the offered call.

Conditions

1. Call Offer is only applied to a station that is in talk status.
2. During a conference or paging, call offer is not activated.
3. System does not support the path reservation mode of standard QSIG specification.

Programming

SYSTEM Networking Basic Attribute (PGM 320)

2.71.2.6 Net Conference

Description

A call to a station on one node is able to conference in a party on any other node. Members of conference must be allowed up to 3 stations of the network.

Operation

To make a Net Conference

1. Press the [CONF] button during conversation through the network call.
   The exiting call is on hold and an ICM dial tone is provided.
2. Make a Net Call to another station of other node.
3. Press the [CONF] button when 3'rd party answers.
   The second call is on hold and an ICM dial tone is provided.
4. Press the [CONF] button again at master station. All parties can make a conversation.

To clear a Net Conference

1. Any station in the net conference hangs up during the conference.
   The net conference will be cancelled and the network path will be cleared.

Conditions

1. Multi-line conference time is not applied on 3 party conference of networking.
2. Standard IP phone can not be a master station of the Net Conference.
   Engineering notice)
   When IP phone sends Invite Setup to Non-IP phone, this message will be rejected.
   When Non IP phone is going to invite IP phone, normal Setup message should send to the IP phone instead of Invite Setup message.

Programming

SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Network Numbering Plan Table (PGM 324)

2.71.2.7 Message Waiting Indication (MWI)

Description

MWI is the same meaning of CLI (Calling Line Indication) message wait service. As usual, in normal ISDN call, a station can leave a message wait for an unanswered station of another node, if CLI message wait is enabled. MWI is indicated by CLI message on station LCD panel. Also, it can be supported by supplementary service (without CLI) when press [MSG/CALLBK] button during hearing ring back tone at Net call.

Operation

To retrieve MWI message

1. Press the flashing [MSG/CALLBK] button and dial 2.
2. MWI contents (CLI number, date and time, the calling count from the same CLI) will be shown on LCD.

700100	
mm/dd hh/mm	CNT:01

3. Press the volume up/down [▲ ▼] button. The previous or the next MWI is displayed.

To delete the current CLI Message and see the next one

1. Press "*" button.
2. Current MWI message is cleared with a confirmation tone and the next MWI message is displayed.

To make a call back

1. Retrieve MWI with pressing the flashing [MSG/CALLBK] button and the volume up/down [▲ ▼] button.
2. Press [SAVE] button.
3. Then system makes a net call according to MWI data.

Conditions

1. MWI is applied to LCD panel installed station only.
2. When system makes a call back according to MWI data, the CO line is selected within the network CO group.

Programming

STATION STATION Attribute (PGM114-BTN10 : CLI Message Wait)

SYSTEM Networking Basic Attribute (PGM 320)
Networking CO Line Attribute (PGM 322)
Network Numbering Plan Table (PGM 324)

2.71.2.8 Net Call Forward – Unconditional

Description

It is possible for a user to remotely forward to another station immediately over the network. Engineering notice).

Currently, iPECS system supports both rerouting and join method by admin programming.

Operation

To activate Net Call Forward

1. Press the [SPEAKER] button and the [FWD] button
2. Dial Net Call Forward code "1", and dial the station number of another system.
   The [FWD] button will be flashing and a confirmation tone will be provided if allowed.
3. Dial the station number that is set Net Call Forward to a station of another node.
   The call is routed to a forwarded station immediately, and the forwarded station is ringing.
4. Answer the ringing at forwarded station.
   The calling and forwarded station can make a conversation.

To deactivate Net Call Forward

1. Press the [FWD] button when it is flashing.
   The [FWD] button will be extinguished.

Conditions

1. If both of served and diverted-to stations are located in the same system, the networking path will be cleared. That is, the forwarded call will be setup as intercom call.
2. System does not check the status of diverted-to station that is in DND, CFW or Empty.

Programming

SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Networking Supplementary Attribute (PGM 321 – BTN 1)
3) Networking CO Line Attribute (PGM 322)
4) Network Numbering Plan Table (PGM 324)

2.71.2.9 Net Call Forward – Busy

Description

It is possible for a user to forward his station remotely when the station is busy. The forwarded destination can be over the network.

Engineering notice).

Currently, iPECS system supports both rerouting and join method by admin programming.

Operation

To activate Net Call Forward

1. Press the [SPEAKER] button and the [FWD] button
2. Dial Net Call Forward code "2", and dial the station number of another system.
   The [FWD] button will be flashing and a confirmation tone will be provided if allowed.
3. Dial the station number that is set Net Call Forward to a station of another node.
   The call is routed to a forwarded station when the called station is busy, and the forwarded station is ringing.
4. Answer the ringing at forwarded station.
   The calling and forwarded station can make a conversation.

To deactivate Net Call Forward

1. Press the [FWD] button when it is flashing.
   The [FWD] button will be extinguished.

Conditions

1. If both of served and diverted-to stations are located in the same system, the networking path will be cleared. That is, the forwarded call will be setup as intercom call.
2. System does not check the status of diverted-to station that is in DND, CFW or Empty.

Programming

SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Networking Supplementary Attribute (PGM 321 – BTN 1)
3) Networking CO Line Attribute (PGM 322)
4) Network Numbering Plan Table (PGM 324)

2.71.2.10 Net Call Forward – No Answer

Description

It is possible for a user to forward his station remotely when the station does not answer within the CFW NO ANS Timer. The forwarded destination can be over the network.
Engineering notice).
Currently, iPECS system supports both rerouting and join method by admin programming.

Operation

To activate Net Call Forward

1. Press the [SPEAKER] button and the [FWD] button.
2. Dial Net Call Forward code "3", and dial the station number of another system.
   The [FWD] button will be flashing and a confirmation tone will be provided if allowed.
3. Dial the station number that is set Net Call Forward to a station of another node.
   The call is routed to a forwarded station when the called station does not answer within a No Answer timer, and the forwarded station is ringing.
4. Answer the ringing at forwarded station.
   The calling and forwarded station can make a conversation.

To deactivate Net Call Forward

1. Press the [FWD] button when it is flashing.
   The [FWD] button will be extinguished.

Conditions

1. If both of served and diverted-to stations are located in the same system, the networking path will be cleared. That is, the forwarded call will be setup as intercom call.
2. System does not check the status of diverted-to station that is in DND, CFW or Empty.

Programming

SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Networking Supplementary Attribute (PGM 321 – BTN 1)
3) Networking CO Line Attribute (PGM 322)
4) Network Numbering Plan Table (PGM 324)

2.71.2.11 Net Call Forward – Busy / No Answer

Description
It is possible for a user to forward his station remotely when the station does not answer within the CFW NO ANS Timer. The forwarded destination can be over the network.
Engineering notice).
Currently, iPECS system supports both rerouting and join method by admin programming.

Operation
To activate Net Call Forward
1. Press the [SPEAKER] button and the [FWD] button
2. Dial Net Call Forward code “4”, and dial the station number of another system.
   The [FWD] button will be flashing and a confirmation tone will be provided if allowed.
3. Dial the station number that is set Net Call Forward to a station of another node.
   The call is routed to a forwarded station when the called station does not answer within a No Answer timer, and the forwarded station is ringing.
4. Answer the ringing at forwarded station.
   The calling and forwarded station can make a conversation.

To deactivate Net Call Forward
1. Press the [FWD] button when it is flashing.
   The [FWD] button will be extinguished.

Conditions
1. If both of served and diverted-to stations are located in the same system, the networking path will be cleared. That is, the forwarded call will be setup as intercom call.
2. System does not check the status of diverted-to station that is in DND, CFW or Empty.

Admin Programming
SYSTEM 1) Networking Basic Attribute (PGM 320)
2) Networking Supplementary Attribute (PGM 321 – BTN 1)
3) Networking CO Line Attribute (PGM 322)
4) Network Numbering Plan Table (PGM 324)

2.71.2.12 CO Transit-In

Description

The incoming DID call can reroute to the net call destination that is in another system.

Operation

1. A DID call is arrived from PX.
   There is no limit for selecting DID conversion type. According to the result of DID conversion, the call will be routed.
2. Network CO line is seized automatically, and the call is transfer to the network destination.
   The destination receives the ringing with CLI from PX.
   The outside user still hears a ring-back tone.
3. Both the outside user and the destination station can make a conversation when the destination station answers the ringing.

Conditions

1. Any timer is not assigned to the transit CO line.
2. Outside caller hears busy tone when a networking path is not available during transit.

Programming

2.71.2.13 CO Transit-Out

Description

The sub-system that has no connection to public network can make a use of the main-system that has connection to public network as well as sub-system. This feature enables a user to maximize use of the network and reduce call costs by routing outgoing calls to the nearest appropriate point on the network. The system should provide sufficient digit translation or string analysis options to enable the switch to route the call correctly.

Operation

1. A station of sub-system seizes CO line.
2. The dummy CO dial tone (in case of PRI real dial tone from PX will be heard) is provided from the main system or the sub-system. According to the CO dial-sending mode of en-block or overlap, the CO dial tone providing system is decided.
3. Dial the telephone number of public network user.
4. The dialed telephone on public network receives the call alert ring, and the station of sub-system hears the ring-back tone.
5. The call connection between the public network telephone and the station of sub-system is established, when the call is answered.

Conditions

1. To use CO transit-out service, the sub-system user must seize the CO line explicitly.
2. The original station's COS is applied for toll restriction.
3. Outside caller hears busy tone when a networking path is not available during transit-out.
4. In the main system, the attendant must have the CO access authority about the public connection of figure.
5. For CO transit out, any code will be available using NET routing table (PGM 324). However, if there is a conflict between NET routing table and system numbering plan, system numbering plan has high priority.
6. If you want to use some code for CO transit out, then its type should be PSTN (PGM 324/BTN1).
7. You can determine that your code is repeated or not by PGM324/BTN7.
8. You must decide that a CO line is connected to PSTN or Network interval at PGM 322 BTN4.
9. Main-system can request a password to sub-system user to apply COS to transit-out call at PGM 324 BTN 12.
10. Sub-system can decide to display dialed digits or not in SMDR because it can contain password (PGM 324 BTN 13).

Programming

SYSTEM 1) Networking CO Line Attribute (PGM 322 / BTN 4)
2) Networking Routing Table (PGM324)

2.71.2.14 BLF Presentation

Description

Stations on one node are able to program a busy lamp appearance of extension on another node. The Busy Lamp Field (BLF) key can also be used to call the remote extension. The BLF presentation can be utilized by the MFIM itself (without BLF manager) or with BLF manager software based on the programming setting.

Operation

With BLF Manager S/W

1. The BLF manager software periodically receives the status of station from whole system.
notice) UDP port will be used to send the status information, and TCP port will be used to send other information.
The BLF manager software sends the broadcast message to whole system when the status is changed.
The network DSS button will be updated according to the PGM 321, 5th (Duration of BLF Status)

Without BLF Manger S/W

According to PGM 324 Btn 6 and 7, System sends BLF information to opposite networked MFIM IP address and port.
The network DSS button will be updated according to the PGM 321, 5th (Duration of BLF Status)

Conditions

1. If BLF manger S/W is used for BLF Presentation, the BLF manager should be installed at one system for completely networked systems.
2. The number of Net DSS can be restricted according to the capability of each system.
3. When a flexible button on a station is assigned as the [NET DSS] button of another system, the system serves as local BLF to indicate the status of the station.
4. CO BLF is not supported, and also ringing signal does not update a status of that station. – ICM / CO / Transfer / CO Recall ring.
5. When a station is in DND mode, the [NET DSS] of DND station is flashing.

Programming

SYSTEM Networking Supplementary Attribute (PGM 321)
- TCP / UDP Port Assign (BTN 2/3)
- BLF manager IP Address Assign (BTN 4)- With BLF manger
- Duration of BLF Status (BTN 5)

Networking Supplementary Attribute (PGM 324)
-DEST MFIM IP Address (BTN 6) – Without BLF manager
-DEST MFIM IP Port (BTN 7) – Without BLF manager

Application

BLF Manager software

2.71.2.15 Do-Not-Disturb (DND)

Description

A call to a station in DND mode can be denied though it is arrived from a station of other systems. The calling party will be heard a busy tone.

Operation

1. Go to DND mode at a station.
2. From a station of other system, dial the station number in DND mode.
   The caller hears a busy tone with LCD display through the network.
   The DND station does not receive any ring signal.

Conditions

When a station is in DND mode, the [NET DSS] of DND station is flashing if BLF manager is activated.

Programming

2.71.2.16 (CAS) Attendant Call

Description

The attendant call from any node can be routed to the centralized attendant.
This call will be queued when all centralized attendants are busy.

Operation

1. Assign Net DSS to the attendant (except system attendant) in PGM 164.
2. System attendant press ATD DND button.
3. Dial attendant code at any station in the system, then the call will be routed to the Net attendant. The system provides ring-back tone to the calling station, and the [Net DSS] button lights.

<u>To assign [ATD DND] button:</u>
[PGM] + {Flex Button} + [PGM] + 83 + [SAVE]

Conditions

1. ICM Call is routed to Net attendant if system attendant press ATD DND button and the next attendant is Net attendant.
2. CO Call can be routed to Net attendant only if DID/DISA destination is ATD and the ring is not assigned to any station.

Programming

STATION Attendant DND Button (PGM 115 – STN PGM 83)
SYSTEM Net Attendant Assign (PGM 164)
Net DSS cannot be assigned to the System Attendant.
DID/DISA Destination (PGM 167)
CO Ring Assignment (PGM 144)

2.71.2.17 Centralized Voice Mail

Description

This function can support that all voice mail occurred in all systems can be recorded in an external VMS.

Operation

Refer to the External Voice Mail function in the system features.

Conditions

1. The centralize VMS should be assigned in slave system, and the number of the centralized VMS should use the representative number of mail access created in master system.
2. The numbering plan including the representative of mail access assigned in master system should be included in the numbering plan of QSIG group in slave system.

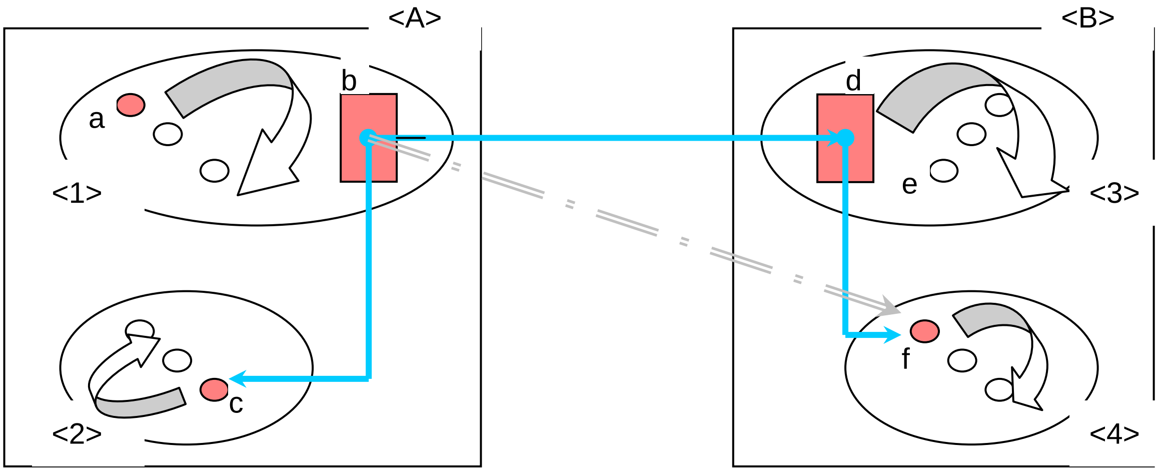
Programming

SYSTEM 1) Network Numbering Plan Table – (PGM 324 – BTN 6)
2) Centralized VMS Assign – (PGM 190)

2.71.2.18 Paging to Networked System

Description

A Station User can initialize a page announcement to other systems by dialing a pre-programmed number or flex button just as if it were a page announcement within the same system.



Operation

1. Enter the Network Numbering plan code to the Network Numbering Plan Table (PGM324) as Net a call in caller system.
2. Enter the Network Paging number and select the page type and destination from the Network Feature Code Table (PGM325) of the called system.
3. Lift the Handset or press the [speaker] button; the System will provides a dial tone.
4. Dial the Network Paging number of the other System,
   OR
5. Press the pre-programmed {NET DSS} button containing the Network Paging number of the other System.
6. The Called party System will receive a digit stream sent by the Calling party; it will be analyzed using the Network Feature Code Table to determine the appropriate destination. If the destination is right, the Called system will distribute the page announcement to the Stations.
7. The [Network CO] button LED will be extinguished when the Net Page Call is cleared.

Conditions

1. Network Page code must be programmed in the Network Numbering Plan Table (PGM 324) in the Caller System.
2. Network Page code and page destination must be programmed in the Network Page Table (PGM 325) in the Called system.
3. Network Page destination could be Internal Page (MFIM100: 1-10, MFIM300/MFIM600/MFIM1200: 1-35), External Page(1-2), All Call Page(Internal, External, All).
4. User will hear an error tone if there is no idle networking path or if the page destination is invalid.
5. User can make a page announcement without lifting the handset.

Programming

SYSTEM

1. Networking Basic Attribute (PGM 320)
2. Networking CO Line Attribute (PGM 322)
3. Network Numbering Plan Table (PGM 324)
4. Network Feature Code Table (PGM 325)

2.72 TRAFFIC ANALYSIS

Description

The iPECS monitors, stores and periodically or upon request outputs various traffic statistics covering system resources. The output from the system can be used to:

- Monitor and evaluate system performance
- Observe usage trends and recommend possible corrective actions,
- Determine possible trunk problems, i.e. blocking level too high, and/or
- Recommend system upgrades.

Attendants enable Periodic Reporting. Once enabled, the system continues to monitor and output the requested report until Periodic Report is disabled. On-demand reports selected by the System Attendant are output only upon request. The Traffic Report is sent to the defined system RS-232 or TCP port.

System resources covered by Traffic Repots are:

- Attendant Traffic Report
- Call Summary Report
- Hourly Call Report
- H/W Unit Usage Summary Report
- CO Summary Report
- Hourly CO Report

Summary Traffic Reports cover one of five Analysis periods selected at time of print:

1. Today’s peak activity hour (within 24 hours)

2. Yesterday's peak activity hour (24 hours prior to Today's activity)
3. Last hour activity
4. Today's total activity
5. Yesterday's total activity.

Operation

System Attendant

To print t the All Summary Traffic Report

1. Press the **[PGM]** button.
2. Dial '0122', the All Summary report code.
3. Select Analysis Period ('1'-'5').
4. Select hour for print ('00'-'23').
5. Press the **[SAVE]** button.

To cancel the periodic All Summary Report

1. Press the **[PGM]** button.
2. Dial '0123', the cancel All Summary report code.
3. Press the **[SAVE]** button.

To print a traffic report

1. Press the **[PGM]** button.
2. Dial the report code, '0121 & 0124' – '0129'.
   - 0121 All Summary Traffic Reports
   - 0124 Attendant Traffic Report
   - 0125 Call Summary Report
   - 0126 Hourly Call Report
   - 0127 Hardware Usage Summary Report
   - 0128 CO/IP Summary Report
   - 0129 Hourly CO/IP Report
3. Press the **[SAVE]** button.

Conditions

1. The Print All Summary Traffic Reports generates the Attendant, Call Summary and CO/IP Summary Traffic Reports.

Programming

SYSTEM
1. RS-232 Port Settings (PGM 174)
2. Serial Port Function Selections (PGM 175)

Related Features

SMDR Call Records

Hardware

device to capture reports

2.72.1 Traffic Analysis, Attendant

Description

The Attendant Traffic Report covers operational statistics for the Attendants. The report outputs periodically or the Attendant requests output of the report for a defined Analysis period. The following is a sample report and description of the report fields.

```
Site Name : abc co
Report Type : Attendant Traffic Report- Today Peak
Date : 06/05/02 08:34
===========================================================================================
ATD  Meas  ----------Calls------------ ---------Time--------------  Speed  ATD
No   Hour  Total Ans   Abdn H   Abd Held  Avail  Talk   Held   Noans  Ans    Type
100  13:00 104    82   22   3   0   18    10:12  14:21  01:23  00:52  00:23  Sys
101  13:00 90     72   15   1   0   11    12:43  30:12  00:54  00:23  00:21  Main
```

Field	Description
ATD No	Attendant Station Number
Meas Hour	(Measurement Hour) Hour data accumulation began
Calls Total	Total number of calls, except group & recalls, routed to the Attendant
Calls Ans	(Calls Answered) Calls answered during the Analysis period
Calls Abdn	(Calls Abandoned) Calls abandoned before answer by the Attendant, does not include calls abandoned while on hold.
Call H-Abdn	(Calls Abandoned from Hold) Calls abandoned while on hold
Calls Held	Number of calls placed on hold by the Attendant
Time Avail	(Time Available) Time attendant was available to handle new calls
Time Talk	Total time the Attendant was active on calls
Time Held	Time Attendant had calls on hold
Time NoAns	(Time No Answer) Average time calls were ringing or in queue for attendant before abandoned
Speed Ans	(Speed of Answer) Average time calls rang before answer by Attendant
ATD type	(Attendant Type) System or Main

Operation

System Attendant

To print the Attendant Traffic Report

1. Press the **[PGM]** button.
2. Dial ‘0124’, the Attendant Traffic report code.
3. Select Analysis Period (‘1’-‘5’).
4. Press the **[SAVE]** button.

Conditions

1. The Peak Hour is the hour when the system has the highest total call volume.

Programming

SYSTEM

1. RS-232 Port Settings (PGM 174)
2. Serial Port Function Selections (PGM 175)

Related Features

SMDR (Station Message Detail Recording)
ACD Statistics Report

Hardware

device to capture reports

2.72.2 Traffic Analysis, Call Reports

Description

Call activity statistics are provided in the Hourly Call and Call Summary Reports.

Hourly Call Report

The Hourly Call Report covers hourly-completed call activity for the selected Analysis period. The report indicates the number of completed calls for each hour during the Analysis period as shown below.

```
Site Name : abc co
Report Type : Attendant Traffic Report- Today Peak
Date : 06/05/02 08:35
===========================================================================================
Anal Hour            # Calls Completed
15:00                 0
14:00                 0
…..
…..
05:00                 211
04:00                 543
Totals Calls          754
```

Operation

System Attendant

To print the Hourly Call Report

1. Press the **[PGM]** button.
2. Dial ‘0126’, the Hourly Call report code.
3. Press the **[SAVE]** button.

Conditions

Programming

SYSTEM

1. RS-232 Port Settings (PGM 174)
2. Serial Port Function Selections (PGM 175)

Related Features

SMDR (Station Message Detail Recording)
ACD Statistics Report

Hardware

device to capture reports

2.72.3 Traffic Analysis, H/W Usage

Description

The Hardware Usage report provides statistics for the system's special Hardware resources such as the VSF as shown in the following sample report.

```
Site Name : abc co
Report Type : Hardware Usage Report- Today Peak
Date : 06/05/02 08:34
===========================================================================
Unit  Num   Anal   Total  Total
Type  Unit  Hour   Req    Denied
VSF   6     09:00  19     0
```

Operation

System Attendant

To print the Hardware Usage Summary Report

1. Press the **[PGM]** button.
2. Dial '0127', the H/W Usage Summary report code.
3. Select Analysis Period ('1'-'5').
4. Press the **[SAVE]** button.

Conditions

Programming

SYSTEM

1. RS-232 Port Settings (PGM 174)
2. Serial Port Function Selections (PGM 175)

Related Features

SMDR (Station Message Detail Recording)
ACD Statistics Report

Hardware

device to capture reports

2.72.4 Traffic Analysis, CO/IP Reports

Description

The CO/IP Traffic Summary and Hourly reports provide statistics on a summary or hourly basis for CO/IP Group activity. The following provides a sample report and description of the major fields in the report.

```
Site Name : abc co
Report Type : CO Group Summary Report- Today Peak
Date : 09/08/02 08:34
===========================================================================================
Peak Hour for All CO 11:00
Grp Num  Anal   Total  Total  Inc    Out    Grp  %    %
No  COs  Hour   Usage  Seize  Seize  Seize  Ovl  ACB  FAO
1   16   10:00  9      6      4      2      0    0    0
20  1    10:00  1      1      0      1      0    0    0
```

Field	Description
Grp No.	CO/IP Group number
Anal Hour	(Analysis hour) hour during the analysis period with peak usage.
Total Usage	Total number of call attempts on CO/IP lines in the Group
Total Seize	Total number of times CO/IP lines in the group were used for any call
Inc Seize	(Incoming Seizures) Total number of incoming calls answered for CO/IP lines in the group.
Out Seize	(Outgoing Seizures) Total number of outgoing calls attempted on CO/IP lines in the group.
ACB	(All COs Busy) Percentage of the time that all CO/IP lines in the group were simultaneously busy.
FAO	(Failed Attempts Outgoing) Percentage of outgoing calls offered to the CO/IP lines in the group that were denied due to All Trunks Busy condition.

Operation

System Attendant

To print the CO/IP Traffic Summary Report

1. Press the **[PGM]** button.
2. Dial '0128', the CO/IP Traffic Summary report code.
3. Select Analysis Period ('1'-'5').
4. Press the **[SAVE]** button.

To print the CO/IP Hourly Traffic Report

1. Press the **[PGM]** button.
2. Dial '0128', the CO/IP Hourly Traffic report code.
3. Select Analysis Period ('1'-'5').
4. Press the **[SAVE]** button.

Conditions

Programming

SYSTEM

1. RS-232 Port Settings (PGM 174)
2. Serial Port Function Selections (PGM 175)

Related Features

SMDR (Station Message Detail Recording)
ACD Statistics Report

Hardware

device to capture reports

2.73 UNIVERSAL NIGHT ANSWER (UNA)

Description

UNA allows a user to be alerted over an external loud bell and answer defined CO/IP calls by dialing a UNA code. While primarily intended for alternate answering during the Night service mode, UNA will also function in other modes providing Universal Answer in all service modes. Calls will appear on the **{CO}**/**{IP}** appearance or a **{POOL}** or **{LOOP}** button. An External Control Contact can be assigned to activate an external Loud Bell to alert users of incoming calls.

iPECS Phones may be assigned a Flex button as a **{UNA}** button.

Operation

iPECS Phone

To assign a Flex button as a {UNA} button

[PGM] + **{FLEX}** + '567' + **[SAVE]**

To access an incoming UNA call

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial '567', the UNA feature code, the oldest UNA call is connected.

Or,

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **{UNA}** button; the oldest UNA call is connected.

SLT

To access an incoming UNA call

1. Lift the handset.
2. Dial '567', the UNA feature code, the oldest UNA call is connected.

Conditions

Programming

SYSTEM 1. Universal Night Answer (PGM 141-Btn 7)

Related Features

LBC (Loud Bell Control)

Hardware

2.74 VMIM/VSF INTEGRATED AUTO ATTD/VOICE MAIL

2.74.1 VMIM/VSF

Description

VMIM

The VMIM (Voice Message Interface Module) includes processing and memory for the iPECS integrated Auto Attendant, Voice Mail and system announcement applications. The memory is employed to store Auto Attendant announcements, voice mail, greetings and messages, and various system prompts. The system prompts (time, date, etc.) are employed by the Auto Attendant and Voice Mail applications as well as other system features. Each VMIM supports 8 channels with 9 hours of voice storage. Each iPECS system, except iPECS-600/1200, may include 2 VMIMs for a total of 16 channels and 18 hours of storage. The iPECS 600/1200 may have up to 8/30 VMIMs for a total of 64/240 channels and 72/270 hours of storage.

VSF

The VSF (Voice Store & Forward) unit, which is equipped in smaller MFIMs, provides the system memory to support the integrated Auto Attendant, Voice Mail and system announcement applications

available in the iPECS. The memory is employed to store Auto Attendant announcements, voice mail, greetings and messages, and various system prompts. The system prompts (time, date, etc.) are employed by the Auto Attendant and Voice Mail applications as well as other system features. Available storage time for the various iPECS models is provided in Table 1.2-1. Approximately 35 minutes of storage is employed for the fixed system prompts.

2.74.2 VMIM/VSF-Auto Attendant

Description

When a call comes into the system through a DID or DISA line, the call may be routed to one of 70 user recorded VMIM/VSF Announcements. An announcement is assigned as a Station Group announcement or as an Auto Attendant announcement with an Audio/Text Table that permits CCR (Caller Controlled Routing). Station Group announcements are played when a call is routed to the group based on definitions in the Station Group Attributes.

For an Auto Attendant Announcement the system will play the announcement and monitor for digits from the connected external party. A CCR Audio/Text Table defines a dialed digit (‘0’ – ‘9’, ‘#’ and ‘*’) to a route. Each single digit is defined a corresponding route:

Station
Station group
Speed Dial number
Page Zone
Voice Mail
VMIM/VSF Announcement

In addition, the system will monitor digits for a station number. If the user dials a station number, the Auto Attendant will complete an unsupervised call transfer to the station.

Operation

System Attendant

To record an Auto Attd Announcement

1. Press the **[PGM]** button.
2. Dial ‘06’, the Message Record code.
3. Dial the VMIM/VSF sequence number.
4. Dial the Announcement number (‘01’ - ‘72’).
5. With Multi Language support, enter the Language number (1~3), the current announcement is played followed by the “Press # to record” prompt.
6. Dial ‘#’.
7. After the beep-tone, record message.
8. Press the **[SAVE]** button to stop recording and save the message.

To delete a recording

1. Press the **[PGM]** button.
2. Dial '06', the Message Record code.
3. Dial the VMIM/VSF sequence number.
4. Dial the Announcement number ('01' - '72').
5. With Multi Language support, enter the language number (1~3), the current announcement is played followed by the "Press # to record" prompt.
6. Dial '#'.
7. Press the **[SPEED]** button during playback to erase message

System

Operation of the CCR Audio Text Tables and Auto Attendant are automatic.

Conditions

1. There are no individual time limits on an Auto Attendant announcement.
2. Up to 500 VSF Announcements may be recorded including Voice Mail user greetings and the 72 VSF announcements.
3. The external caller may experience ring-back tone before playback of a VMIM/VSF announcement.
4. The System Attendant must "Save" a recording before returning to the on-hook state, otherwise, the existing recording is used.
5. To record or delete an Auto Attendant message, all of the VSF or VMIM channels must be in the idle state.
6. The external caller may dial at any time during an Auto Attendant announcement and must dial prior to the expiration of the CCR Analysis timer.
7. If the external caller dials an invalid selection or station, the system will play the 'Invalid Entry' prompt and allow re-entry using the DISA Retry Counter.
8. If the external caller dials more than a single digit, the call is routed based on the System Numbering Plan.
9. Calls answered by an Auto Attd (CCR) Announcement are interactive DISA calls and are subject to conditions of a DISA call.
10. The '*' digit is reserved in the Audio/Text Tables to repeat the current or previous Auto Attd announcement.
11. The '#' digit is reserved for callers to access their Voice Mailbox remotely.
12. A CCR Announcement may be programmed to disconnect after playing.
13. The Auto Attd Announcement feature is supported for DISA and DID calls.
14. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.
15. System announcement number 71, if recorded, is played as MOH.
16. System announcement number 72 is for Multi-Language support only.

Programming

STATION 1. VSF Access (PGM 113-Btn 2)

CO/IP	1.	DID Service Attributes (PGM 145)
STATION GROUPS	1.	Station Groups (PGM 190)
	2.	Station Group Attributes (PGM 191)
SYSTEM	1.	DISA Retry Counter (PGM 160-Btn 5)
	2.	DID/DISA Destination (PGM 167)
	3.	CCR Analysis Timer (PGM 180-Btn 13)
	4.	VSF User Record Timer (PGM 181-Btn 3)
	5.	VSF Valid User Message Timer (PGM 181-Btn 4)
TABLES	1.	CCR Audio Text Tables (PGM 228)
	2.	Flexible DID Table (PGM 231)

Related Features

Station Groups
Automatic Call Distribution
Direct Inward System Access (DISA)
Direct Inward Dial (DID)
Remote Message Retrieval
Multiple Language Support
MOH (Music-On-Hold)

Hardware

VSF and/or VMIM

2.74.3 VMIM/VSF Voice Mail

2.74.3.1 Message Storage

Description

When a station activates Call Forward to the VMIM/VSF Group, a call is transferred to a VMIM/VSF mailbox or a transferred call recalls the VMIM/VSF, the call is handled by the iPECS Voice Mail application. The caller connects to the called station's User Greeting followed by beep tone.

The remote caller can record a message and hang-up or dial '*' for further options. At disconnect, the VM application stores the message in the called user's voice mailbox and activates the Message Waiting indication at the user's station. If VM back up is assigned, the back-up station (Phontage or UCS Client) is also notified.

Operation

Remote Caller

To leave a voice message after Greeting and beep leave the message

1. Hang up to quit recording or dial '*' for further options.

Conditions

1. Two timers are provided to control voice message length. The Valid User Message Timer establishes the minimum voice message length. Voice messages shorter than this timer are not stored. The VSF User Record Timer establishes the maximum voice message length. When the VSF User Record Timer expires while recording a voice message, confirmation tone is heard and the message is saved for destination station.
2. If all the VMIM/VSF channels are in use, Ring Back tone is provided until a VMIM/VSF channel is available.
3. All stations including an SLT can leave and receive voice messages.
4. Individual User Greetings and Voice Mails are protected from loss of AC power.
5. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.

Programming

STATION

1. VSF Back-up Station (PGM 112-Btn 23)
2. VSF Access (PGM 113-Btn 2)
3. Direct Transfer to Mailbox (PGM 120-Btn 6)
4. Station Call Forward Timer (PGM 123)

SYSTEM

1. Call Forward No Answer Timer (PGM 181-Btn 1)
2. VSF User Record Timer (PGM181-Btn 3)
3. VSF Valid User Message Timer (PGM 181-Btn 4)

Related Features

Call Forward
Station Message Wait/Call Back
VMIM/VSF Voice Mail
Call Transfer, Voice Mail

Hardware

VSF and/or VMIM

2.74.3.2 Message Retrieval

Description

A user can access their Mail Box locally by placing a call to the VMIM/VSF Voice Mail group or, from an iPECS Phone, by pressing the **[MSG/CALLBK]** button, or by pressing a **{VMAILBOX}** button while off-hook receiving Intercom dial tone.

Prompts are then received to guide the user in the Voice Mailbox operation. The user must enter a Mailbox number, generally the station number, and a corresponding password in response to the "Request for Mailbox number" *("Please enter your Mailbox number.")* and "Request for Password" *("Please enter your password code.")* prompts.

If the user enters valid and matching Mailbox and password numbers, the "Number of Messages" prompt *("You have xx new messages. You have yy saved messages.")* is received. At this point, the user also receives the "VM long option prompt" *("To play new messages, press one, to play saved messages, press two, to set greeting or password, press eight, to disconnect, press pound, Press 0 for the operator, Press nine to hear this message again.")*.

When the user responds by dialing 1, the first new message is played. At the end of message playback, the "New Message option" prompt *("To replay message, press one, to listen to the next message, press two, to delete message, press three, to forward message, press four, to call the sender, press five, to skip message, press six, to return to main menu, press nine.")* is played. This process is repeated until the last new message is played and the "No Message" prompt *("No Messages")* is played.

When the user dials 2 in response to the "Number of Messages" prompt, the oldest saved message is played. At the end of the message, the "Old Message option" prompt *("To replay message, press one, to listen to the next message, press two, to delete message, press three, to forward message, press four, to call the sender, press five, to return to main menu, press nine.")* is played. This process is repeated until the last new message is played and the "No Message" prompt *("No Messages")* is played.

In addition to the options indicated in the prompt, a user can record a memo, which is attached to the current voice mail by dialing the digit 7. The current voice mail and memo can then be forwarded to other iPECS users.

When the user dials 9 in response to the "Number of Messages" prompt or during or at the end of a message the "VM long Options" prompt is played.

Operation

iPECS Phone

*To assign a **{VMAILBOX}** Flex button;*

[PGM] + **{FLEX}** + VM group + Mail-box (station) number + **[SAVE]**

To retrieve Voice Mail locally

1. Lift the handset or press the **[SPEAKER]** button
2. Press **[MSG/CALLBK]** button. The message contents summary is shown as below.

ST	CL	VS	VM	FS	MS
001	001	005	006	001	004

3. Dial digit '3' to select VMIM/VSF Messages
4. Press the **[MSG/CALLBK]** to receive the "Mail Box & Password" prompts sequentially.
5. Dial the Mail Box and corresponding password to receive the "Number of Messages" prompt.
6. Dial desired option code.
7. At completion of session, hang-up to return to idle.

Or,

1. Lift the handset or press the **[SPEAKER]** button
2. Press **{VMAILBOX}** button.
3. Dial the Mailbox password to receive the "Number of Messages" prompt.
4. Dial desired option code.
5. At completion of session, hang-up to return to idle.

To attach a memo to the current voice message;

1. During or after the New or Old Message option prompt, dial '7'.
2. At the beep, record the memo.
3. Dial '*' to stop recording and store the memo.
4. During or after the New/Old option prompt, dial 4 to forward the message and memo.

SLT

To retrieve Voice Mail locally

1. Lift the handset.
2. Dial the Voice Mail Group to receive the "Mailbox & Password" prompts sequentially.
3. Dial the Mailbox and corresponding password to receive the "Number of Messages" prompt.
4. Dial desired option code.
5. At completion of session, hang-up to return to idle.

To attach a memo to the current voice message;

1. During or after the New or Old Message option prompt, dial '7'.
2. At the beep, record the memo.
3. Dial * to stop and store the memo.
4. During or after the New Old option prompt, dial '4' to forward the message and memo.

Conditions

1. If no new/old messages are available, pressing '1' or '2', is an invalid operation and the user receives the "Invalid Entry" prompt or "No Message" prompt.

2. If the dialed number is not recognized, the "Invalid Entry" prompt is played. After the second invalid entry, the user is disconnected.
3. The user may dial digits at any time during a voice mail playback, system prompt or silence. The user must dial a digit in response to a system prompt within the CCR Analysis timer or the system will disconnect and return error tone.
4. Messages can be retrieved in either a FIFO (First in First out) or LIFO (Last in First out) order to meet the desire of each user.
5. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.
6. If user set auto save new message and hear new message, it's saved automatically.

Programming

STATION
1. Retrieve Message Order (PGM 113-Btn 13)
2. Auto save new message(PGM 161-Btn 24-11)

Related Features

Message Retrieval Options
Remote Message Retrieval
Multiple Voice Mailbox Support

Hardware

VSF and/or VMIM

2.74.3.3 Remote Message Retrieval

Description

The system permits remote users access to their mailbox. After accessing the VMIM/VSF Voice Mail, operation follows the local procedures.

Operation

Remote Caller

To access Voice Mailbox from a remote location
1. Lift the handset.
2. Dial the telephone number of a DISA assigned CO line assigned for answer by a VMIM/VSF Auto Attd.
3. Upon answer, dial '#' to receive the "Request for Mail Box number" prompt.
4. Follow local access procedures.

Conditions

1. The conditions associated with Message Retrieval and Message Retrieval Options apply.
2. The conditions associated with DISA apply.
3. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.

Programming

Related Features

Message Retrieval Options
VMIM/VSF-Auto Attendant
Message Retrieval

Hardware

VSF and/or VMIM

2.74.3.4 Message Retrieval Options

Description

The user may dial the digit 9 to receive the “VM long Options” prompt while in the Voice Mailbox, including during or after a voice message or system prompt except when an option has been selected that requires user dialing. The later may occur when the user selects Message Retrieval Option 1/2 (Play New/Saved Message), 7 (Cancel or Forward message, for Remote Access Only) or 8 (Mail Box settings). The “VM long Options” prompt is:

"To play new messages, press one, to play saved messages, press two, to set station forwarding, press seven (This option is available only for remote access), to set greeting or password, press eight, to disconnect, press pound, Press 0 for the operator, Press nine to hear this message again."

The VMIM/VSF Voice Mail will respond to incoming digits as shown in the following table.

Digit	Function
1	Play New Msg
2	Play Saved Msg
7	Set Cancel/Fwd, available only for remote access
8	Mailbox Setting, "Mailbox Settings" prompt
9	VM Long options
#	Drop, "Goodbye”
0	Attd Call, Call to System Attendant.

Operation

iPECS Phone

To access a Message Retrieval option

1. At any time after the “Number of Messages” prompt, dial a Message Retrieval Option digit. The system initiates the selection providing any needed prompts.

SLT

To access a Message Retrieval option

1. At any time after the “Number of Messages” prompt, dial a Message Retrieval Option digit. The system initiates the selection providing any needed prompts.

Conditions

1. The user must begin dialing within the CCR Analysis timer in response to a system prompt. If the timer expires, the system will disconnect the call and the user will receive error tone.
2. When the “call sender” option results in an external call, dialing restrictions will be applied based on the Station COS.
3. If the user remains off-hook after a call placed through the voice mail is complete, the user will be returned to the previous place in the Voice Mailbox. If the user hangs up, the VMIM/VSF will recall the user and, upon answer will play “Request Mailbox Number” prompt.
4. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.
5. If you set auto save next message ON, it’s saved automatically after hearing the new message.

Programming

SYSTEM 1. AUTO SAVE NEXT MSG (PGM 161-Btn 24 - 11)

Related Features

Message Retrieval
Remote Message Retrieval
Voice Mailbox Settings
Class of Service

Hardware

VSF and/or VMIM

2.74.3.5 E-Mail Notification

Description

With the VSF or VMIM, the system stores the voice message and sends an e-mail to the e-mail address associated with the station as notification of the new e-mail. The voice message can be attached to the e-mail as a wav file.

Operation

System

System automatically sends e-mail to notify user of new voice message.

Conditions

1. The voice message is stored in the VMIM/VSF as well as being attached to the e-mail. The voice message must be expressly deleted from the VMIM/VSF even if the e-mail is deleted.
2. The e-mail is sent to the address assigned for the station with the "sender" address defined for the VMIM/VSF. Note the latter is required, as many e-mail servers will reject anonymous e-mails.
3. The e-mail address for the VMIM/VSF and the station is defined under the Web Admin.
4. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.
5. The Voice message can be attached to the e-mail notification as a .wav file, if the Attach Message option (**PGM** 111) is enabled. If disabled, the notification e-mail does not include an attached wav file.
6. If the below E-mail program is duplicated with each other, Station program have prority.

Programming

STATION

1. VSF MSG – SMTP Mail Server Address.
2. VSF MSG – Receiver Mail Address.
3. VSF MSG – SMTP Mail Server ID.
4. VSF MSG – SMTP Mail Server P/W.
5. VSF MSG – Add Message .
6. VSF MSG – Delete Message After Add.
7. VSF MSG – SMTP Security.
8. VSF MSG – SMTP Port.
9. VSF MSG – Sender Mail Address.

BOARD

1. Sender Mail Address (PGM 132)

SYSTEM

1. VSF/VMIM SMTP Port.

Related Features

VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

VSF and/or VMIM

2.74.3.6 Voice Mail Back-up Station

Description

With the VSF or VMIM, an iPECS soft-phone (Phontage and UCS Client) user receives notification of new messages for assigned stations. The soft-phone will indicate the total messages for the assigned stations and the soft-phone. The soft-phone user can download the messages for other stations to the hard drive of the soft-phone PC and, using the soft-phone application, manage the messages on the hard drive. If enabled, the soft-phone user may delete voice messages from the VMIM/VSF memory.

Operation

System

System automatically notifies back-up station of new messages

Phontage or UCS Client

See Phontage or UCS Client Guide

Conditions

1. Voice messages are stored in the VMIM/VSF and must be expressly deleted. Deleting messages from the soft-phone hard drive does not delete the message from the VMIM/VSF memory.
2. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.
3. It is not possible to backup VSF Messages to Phontage PC when Phontage is connected Remotely in R/NAPT mode.

Programming

STATION

1. VSF Back-up Station (PGM 112-Btn 22)
2. VSF Back-up Delete (PGM 112-Btn 23)

Related Features

VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

VSF and/or VMIM

2.74.3.7 Voice Mailbox Settings

Description

The user can program the Mailbox settings for their mailbox including a security password and a greeting. When a user presses '8' while retrieving messages, the "Mailbox Setting" prompt, *("To edit your greeting, press one, to edit you password, press two. To return to main menu, press nine") is played*.

Operation

To program Mailbox settings while "in" the Voice Mailbox

1. Press '8', for Mailbox settings, the "Mailbox Setting" prompt is received.

for password

1. Dial '7' and receive the "Password Entry" prompt ("Please enter your new password and press pound when finished.").
2. Dial new password.
3. Press '#' and receive the "Reenter Password" prompt ("Please re-enter your password to confirm and press pound when finished.").
4. Dial new password again.
5. Press '#' and receive "Password Confirmation" prompt *("Your password is saved.")*.

for greeting

1. Dial '4' and receive "Greeting Option" prompt ("To listen to your current greeting, press five to record a new greeting, press seven to return to the main menu, press nine.").
2. Dial '5', to hear your greeting.

Or,

1. Dial '7' and receive "Record Greeting" prompt, ("At the tone, record your new greeting, press # when done.").
2. After beep, record greeting speaking in a normal voice.
3. Press '#' and receive "Greeting Confirmation" prompt ("Your greeting is saved.").

for "Mailbox Setting" prompt

1. Dial '9' and receive "Mail Box Setting prompt" ("To edit your greeting, press one, to edit you password, press two. To return to main menu, press nine").

Conditions

1. If the user is external, the user must begin dialing within the CCR Analysis time, if not the call is released.
2. If the dialed number is not recognized, the "Invalid Entry" prompt is played.
3. The user must assign a password (Authentication code up to 12 digits) before access to the mailbox will be allowed. Note that a greeting need not be recorded.
4. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.

Programming

Related Features

Message Storage
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VSF and/or VMIM

2.74.3.8 Call Forward from VM

Description

External users can activate or deactivate Call Forward for their station. Pressing ‘7’ while retrieving messages will return the "Mailbox Set Forward" prompt.

Operation

To activate Call Forward while in the VM;

1. Press ‘7’, for Mailbox set forward, the “Mailbox Set Forward” prompt is received.
2. Dial ‘1’ and receive the “Password Entry” prompt (“Please enter the number to forward to ...").
3. Dial Station Number as follows:
   - To forward to another station, dial the station number.
   - To forward calls Off net, dial ‘*’ and enter station speed number. If the station Speed bin is valid, the confirmation announcement “forwarded to station ('xxx')” or “forwarded to speed bin number (yyyy)” is played.

To deactivate Call Forward

1. Press ‘7’, for Mailbox set forward, the “Mail Box Set Forward” prompt is received.
2. Dial ‘2’ and receive the “Station forwarding is canceled” prompt.

To return to the Main menu

1. Dial ‘9’ and receive the “Mailbox Settings” prompt.

Conditions

1. If the user is external, the user must begin dialing within and dial subsequent digits within the VSF Inter-Digit Timer. If not, the call is released.
2. This Mailbox Set Forward is only available for external users.
3. In Admin, VSF is used as the single reference for a parameter that applies to both the VSF and the VMIM.

Programming

Related Features

Message Storage
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VSF and/or VMIM

2.74.3.9 Outbound Message Notification

Description

The VSF and VMIM are able to dial an external number to notify a user of a new voice message. The system employs the mobile extension number registered for the station receiving the message. When a caller leaves a message with notification configured, the system places a call to the registered mobile extension. When the user answers, the extension prompt is played followed by the new message prompt, ("You have xx new messages."). The new message prompt indicates the number of unheard messages.

The user must listen to the new message to confirm the notification. If the user takes no action within the CCR Inter-digit timer or hangs-up, the call is disconnected and the system will retry the call after the retry timer expires, until the user listens to the message or the number of attempts reaches the retry counter. If the user does not answer, the ISDN or VoIP connection times out or disconnects before answer, or is busy, the system disconnects the notification and will retry the call after the retry timer. The system will retry the notification until the notification is successful or the number of call attempts reaches the Retry count.

Operation

Operation of message notification is automatic when configured.

Condition

1. Outbound notification over a PSTN line is not available.
2. Caller Id will be the external caller who left the message or, for messages from another station, Caller Id will be the station receiving the message.
3. If VSF/VMIM Notify is changed to 'Not Use', any existing notification will be terminated after the initial notification call.
4. For proper operation, the Station COS and CO Group access for the station must be such as to allow the notification call.
5. The destination of the notification is the Mobile telephone number assigned in PGM 236.
6. If all lines in the assigned CO group are busy when the system attempts to place the notification call, the System will continuously try to seize CO line until a line is successfully seized.
7. The Retry counter is incremented after the system access the CO line for notification.
8. The Retry count is from 1 to 9; the retry interval is from 1 to 3 minutes.
9. If a new message is logged before answer of the notification call, the message will be available to the user and a new notification is not invoked. If a new message is received after answering the notification call, the System will invoke another notification call. The user will receive the notification after returning to idle.

Programming

TABLES

1. Tel Number (PGM 236-Btn 6)
2. VSF/VMIM Notify (PGM 236-Btn 7)
3. Notify Retry (PGM 236-Btn 8)
4. Retry Interval (PGM 236-Btn 9)

Related Features

Mobile Extension
Message Retrieval
Remote Message Retrieval
Message Retrieval Options

Hardware

VSF and/or VMIM

2.74.4 System Voice Memo

Description

This feature provides several general Voice Memos to provide the system time and date as well as station number, and settings over the iPECS Phone speaker or the handset for SLTs.

Operation

iPECS Phone

To hear Date & Time Prompt

1. Dial feature code,

'661' for IPECS-Micro & 50 & 100
'*661' for all other systems
Announcement for date & time is heard, "Date is May 2nd. Time is xx:xx pm".

To hear Station Number Prompt

1. Dial feature code,
   '662' for IPECS-Micro & 50 & 100
   '*662' for all other systems
   the station number announcement for station is heard, "This is station 150".

To hear Station Settings

1. Dial feature code,
   '663' for IPECS-Micro & 50 & 100
   '*663' for all other systems
   status for the station is reported. Items reported are as follows:
2. Station IP Address
   Station Mac Address
   Station ICM Mode (Handsfree/Tone/Privacy)
   Listed message x (x: number of all message waiting)
   Wake-Up Time (hh:mm)
   Do not disturb
   Forwarded to station xxx
   Forwarded to speed bin xxx
   Queued CO/IP xx
   Locked (temporary COS change)
   COS x

SLT

To hear Date & Time Prompt

1. Lift the handset.
2. Dial feature code
   '661' for IPECS-Micro & 50 & 100
   '*661' for all other systems
   announcement for time is heard, "Date is May 2nd.Time is xx:xx pm".

To hear Station Number Prompt

1. Lift the handset.
2. Dial feature code
   '662' for IPECS-Micro & 50 & 100
   '*662' for all other systems
   announcement for station is heard, "This is station 150".

To hear Station Settings

1. Lift the handset.
2. Dial feature code,
   663 IPECS-Micro & 50 & 100
   *663' for all other systems
   status for station is reported. Items that will be reported are as follows:

Station number
Station IP Address
Station Mac Address
Station ICM Mode Handsfree/Tone/Privacy)
Listed message x (the number of all message waiting)
Wake-Up Time (hh:mm)
Do not disturb
Forwarded to station xxx
Forwarded to speed bin xxx
Queued CO/IP xx
Locked (temporary COS change)
COS x

Conditions

1. For station status, items from "Listed message x" to "COS x" will be not be announced if not active.

Programming

STATION 1. VSF Access (PGM 113-Btn 2)

Related Features

VMIM/VSF

Hardware

VSF and/or VMIM

2.75 WAKE-UP ALARM (ONE TIME WAKE UP ALARM)

Description

This feature allows a user or Attendant to set a wake-up time or desired time to be alerted. When the time is reached, the system will signal with an audible and visual signal.

Operation

Attendant

To register Wake-Up

1. Press the **[PGM]** button.
2. Dial '044', Attendant Station Program code.
3. Dial the desired station range, for a single station, enter an '*' in place of the second station number.

4. Dial 2-digit hour and 2-digit minute for alerting.
5. For a daily (repeating alarm), dial '#'.
6. Press **[SAVE]** button.

To erase Wake-Up

1. Press the **[PGM]** button.
2. Dial '045', the Attendant Station Program code.
3. Dial the desired station range, for a single station, enter an '*' in place of the second station number.
4. Press **[SAVE]** button.

iPECS Phone

To register Wake-Up

1. Press the **[PGM]** button.
2. Dial '41', the set Wake-up code.
3. Dial 2-digit hour and 2-digit minute for alerting.
4. For a daily (repeating alarm), dial '#'.
5. Press **[SAVE]** button.

To stop the alarm notification

1. Lift the handset or press **[SPEAKER]**.

To erase Wake-Up

1. Press the **[PGM]** button.
2. Dial '42', the erase Wake-up code.
3. Press **[SAVE]** button.

SLT

To register Wake-Up

1. Lift the handset.
2. Dial '561', the SLT Programming code, confirmation tone is heard.
3. Dial '41', the set Wake-up code.
4. Dial 2-digit hour and 2-digit minute for alerting.
5. For a daily (repeating alarm), dial '#'.
6. Hook flash, conformation tone is provided.

To stop the alarm notification

1. Lift the handset.

To erase Wake-Up

2. Lift the handset.
3. Dial '562', the SLT Programming code, confirmation tone is heard.
4. Dial '42', the erase Wake-up code.
5. Hook flash, a conformation tone is provided.

Conditions

6. In cases where a remote device is installed, the time display in the remote terminal is adjusted by the Device Zone assignment. Thus, the Wake-up alarm time is based on the time displayed on the phone and not the system time.
7. When receiving a wake up signal, lifting the handset will return MOH.
8. The Wake-up alarm Ring signal is 30 seconds On/90 seconds Off repeated 3 times. If no action is taken by the user, the ring signal is given to the Attendant with a display designating the station number that did not respond.
9. Time (hh:mm) must be entered in the Military 24-hour format.
10. The daily alarm will reset and repeat each day until erased (cancelled). The One-time alarm will reset and cancel automatically.

Programming

Related Features

Hardware

2.76 WAKE-UP ALARM (FIVE TIME WAKE UP ALARM)

Description

If [PGM161 – New 5 Wake Up Usage] is set to ON, then new wake-up feature is enabled.
Each station can have and set up to 5 wake-up times. Each wake-up time is identified using an id (1-5).
Each wake-up time has a wake-up type:

1	YY/MM/DD	Alarm will be activated only one time in day specified by YY/MM/DD.
2	Mon – Fri	Alarm will be activated from Monday to Friday.
3	Mon - Sat	Alarm will be activated from Monday to Saturday.
4	Mon - Sun	Alarm will be activated every day.

Operation

<u>*To register wake-up time alarm from a Station, perform the following steps:*</u>

1. Press the [TRANS/PGM] button.
2. Dial [4] [1].
3. Enter wake-up time id: [1] … [5].
4. Dial the desired 2-digit hour (24-hour mode), then 2-digit minute for alerting.
5. Enter wake-up type: [1] … [4].

6. If wake-up type is 1 (YY/MM/DD), enter wake-up date: 2 digits for year, 2 digits for month and 2 digits for day.
7. Press the [HOLD/SAVE] button.

To erase a wake-up time from a Station, perform the following steps:

1. Press the [TRANS/PGM] button.
2. Dial [4] [2].
3. Enter wake-up time id: [1] … [5].
4. Press the [HOLD/SAVE] button.

1.

To register a wake-up time alarm from the Attendant Station, perform the following:

1. Press the [TRANS/PGM] button.
2. Dial [0] [4] [4].
3. Dial the station range to be alerted. If a single station is to receive the alarm, enter ＊ instead of a second station number.
4. Dial the desired 2-digit hour (24-hour mode), then 2-digit minute for alerting.
5. Enter wake-up type: [1] … [4].
6. If wake-up type is 1 (YY/MM/DD), enter wake-up date: 2 digits for year, 2 digits for month and 2 digits for day.
7. Press the [HOLD/SAVE] button.

To erase a wake-up alarm from the Attendant Station, perform the following Steps:

1. Press the [TRANS/PGM] button.
2. Dial [0] [4] [5].
3. Dial the station range that was to be alerted. If a single station is to receive alarm, enter ＊ instead of a second station.
4. Press the [HOLD/SAVE] button.

Conditions

1. When register or erase wake-up time from a Digital station, it can use [VOL UP] and [VOL DOWN] buttons to scroll between wake-up times.
2. When register wake-up time from a Digital station, it can use [#] button to scroll between wake-up types.
3. When register wake-up time from Attendant Station, wake-up information will be saved in a free wake-up entry, for each station from station range. If one station has not any free wake-up entry, then the wake-up time from Attendant Station will not be saved.
4. When erase wake-up time from Attendant Station, all wake-up information will be erased.
5. When [PGM161 – New 5 Wake Up Usage] : changed from OFF ➔ON or from ON ➔OFF , all wake-up times for all stations are erased and current wake-up times are canceled.
6. When wake-up time is active in current day, a *(star) will signalize near clock displayed on Digital station LCD. If there is no wake-up time enabled in current day, then there is no signalization on Digital station LCD.
7. After finalizing wake-up call, wake-up time is not be automatically erased. Including wake-up time with YY/MM/DD type. A wake-up time can be erased only manually.

8. After finalizing wake-up call (with answer or not answered), a new wake-up time will be enabled, if it exist for current day.
9. Wake-up call functionality is similar with old wake-up feature.
10. Wake-up year can be from 2000 to 2031.

Programming

PGM161 : ENABLE ENHANCED WAKEUP (ON/OFF)

Related Features

Hardware

2.77 Zone Holiday Assignment

Description

The Administrator can assign Holiday Zones (up to 40) and Vacation Zones (up to 5 for each Zone). Devices in a Zone follow the holiday and vacation assignments of the zone. When the date of a zone is set to Holiday or Vacation, the zone will operate in the programmed ring mode.

Operation

Operation of Zone Holiday Assignment is automatic when programmed.

Conditions

1. When the Tenancy group Ring mode for Flexible DID/ICLID function (PGM231) is programmed, the priority is higher than the Zone Holiday Assignment (PGM444).
2. Zone Holiday Assignment has a higher priority than the Attendant ring mode.
3. It is recommended to use the same zone for each Tenancy group CO and station.

Programming

SYSTEM 1. Zone Holiday Assignment (PGM 444)

Related features

Hardware

2.78 Remote Device Zone Management

Description

Remote devices, in particular those not reachable by the system, are managed by grouping devices by various characteristics in a Zone. Placing devices into Zones simplifies management allowing definition of common characteristics to the devices within the zone. Zone attributes include:

- Nation code
- Language
- Page area
- Time-zone, GMT or system
- Holiday ring modes
- RTP relay
- RTP Relay group.

Operation

Zone operation is automatic once configured.

Conditions

1. Zone attributes which may affect ringing, such as Holiday and Vacation settings, have lower priority than other ring assignments such as Tenancy group ring mode established for DID lines.
2. Zone Attributes do not provide adjustment of time for DST.
3. It is recommended to assign CO/IP lines and stations of a Tenant group in the same Device Zone.
4. Wake-up time is based on the time displayed in the station's LCD.

Programming

Zone Data	Device Zone Number (PGM436-Web only)
	Device Zone Attributes (PGM 437-Web Only)
	Zone Access & Page Relay (PGM 438-Web only)
	Zone Attributes (PGM 439-Web only)
	Zone RTP Relay Group (PGM 440-Web only)
	Inter-zone Attributes (PGM 441-Web only)
	Zone Holiday Assignment (PGM 444)

Related features

Centralized Control TNET

Hardware

2.79 ZONE CALL AND CONNECTION NUMBER LIMIT

2.79.1 Zone Outgoing CO call Number Limit

Description

Every system has the limited CO lines and all devices are sharing to use CO lines. The number of CO lines may be smaller than the number of other devices. If some devices use all the CO lines, the other devices can't call to outside of system. To prevent this situation and to increase the common usage of the CO lines, outgoing CO calls are limited to zone maximum outgoing CO call number.

Zone maximum outgoing CO call number can be set from 0 to maximum CO line capacity. The default value of zone maximum outgoing CO call number is 0 and it means there is no limit to zone outgoing CO call.

Operation

Operation of this feature is automatic based on zone maximum outgoing CO call number.

Conditions

1. If maximum outgoing CO call number is set to 0, zone outgoing CO call will not be limited.
2. If maximum outgoing CO call number is set to non-zero value and current zone outgoing CO call number is more than maximum value, the outgoing co call trying station will play error tone and display "OUTGOING CALL LIMIT OVER" on the LCD.
3. Networking call is also processed as an outgoing CO call.
4. System counts emergency call in current outgoing CO call number, but no limit procedure is applied.

Programming

ZONE DATA	Zone Attributes (PGM 439-Web only)

Related Features

Hardware

2.79.2 Inter Zone Connection Number Limit

Description

If some devices exist very far from the other devices, it might cause to the bandwidth limitation. Those devices must be controlled the number of connections according to the bandwidth limitation for the voice quality. Inter zone connections are limited to zone maximum total inter zone connection number and maximum inter zone connection number between zones.

Zone maximum total inter zone connection number and maximum inter zone connection number can be set from 0 to maximum station capacity. The default values of both numbers are 0 and it means there is no limit to inter zone connections.

Operation

Operation of this feature is automatic based on zone maximum total inter zone connection number and maximum inter zone connection number between zones.

Conditions

1. If zone maximum total inter zone connection number is set to 0, zone total inter zone connection will not be limited.
2. If maximum inter zone connection number between zones is set to 0, inter zone connection between such zones will not be limited.
3. If zone maximum total inter zone connection number is set to non-zero value and current zone total inter zone connection number is more than maximum value, the connection will be released, error tone will be played and the station related with the connection will display "INTER CONN LIMIT OVER" on the LCD.
4. If maximum inter zone connection number between zones is set to non-zero value and current inter zone connection number between zones is more than maximum value, the connection will be released, error tone will be played and the station related with the connection will display "INTER CONN LIMIT OVER" on the LCD.
5. As the connection limit is processed at the moment of the connection establishment, if the limit is over while incoming ring is answered, the call will be released immediately after answering and error procedure will be activated.
6. If the recording connection for call recording is over the limit, the call will be recorded without content.

Programming

ZONE DATA	Zone Attributes (PGM 439-Web only) Inter Zone Attributes (PGM 441-Web only)

Related Features

Hardware

2.80 SYSTEM CALL ROUTING

Description
System can change destination automatically or manually by condition of time , days , CLI, CPN, CO, Zone and Tenancy Group number about incoming calls. If scenarios are set, incoming calls are routed to the destination(like as 231 Flex Did destination) by the condition.

Item of Condition Table

- Caller ID
- Called Num
- Days and time(Start Day and End Day, weeks, start time and end time).
- Destination(Type and Value)
- Scenario Priority
- Scenario Active or Not.
- Scenario Voice Mail box.
- Scenario COS.
- Scenario Disa Active or Not.
- Scenario Tenancy Group number.
- Scenario Zone.
- Scenario Start CO and End CO.
- Scenario Group : this is used for scenario group by attendant.

Conditions

1) Scenario Active should be set ON and Day condition choisen a day or everyday for operation.
2) Even though System is Day/Night/Timed mode, it works by system call routing scenario. And so, System call routing is hightest priority than others.
3) We recommend that Co and Station are same tenancy group and zone.
4) If Start time is 0700 and End time is 1900, it works from AM 07:00 to PM 07:00.
5) The total numbers of table are 16(MFIM1200 : 32) and have 10 scenario respectively. And so you can set maximum 160(MFIM1200 : 320) scenarios.
6) VMID should be Station number that is virtual or not. Virtual means that Station number is exist but it has not Hard Phone. VMID is used when Destination Type is Hunt and Destination is VSF hunt group.
7) If you set Disa Active ON and Destination is VSF 1, CCR will work after announcing 1.
8) If VSF is 0 and Disa Active ON, you can hear Disa dial tone.
9) If you choose ATD STN for the type of Caller ID, it works rerouting to STA,Hunt,Mail Box,Network call and SPEED instead of ATD for internal attendant call in this condition.
10) Scenario group is only used as manual setting by attendant
11) If attendant want to use manual SCR using Scenario group, you should do as following.
Operation>
    1. Press DND in system attendant.

2. Choice 5.
3. Select Scenario (01 ~ 16/32(if MFIM1200)) – This is Scenario Group number in ADM 251.
4. Press SAVE button.

Programming

SYSTEM System Call Routing (PGM251-Web only)
Station Icm Group (PGM 125)
CO tenency group (PGM 141-B10)

Related features

Hardware

2.81 CO CALL REROUTING

Description

System can reroute incoming call to CO. If called number matched with compare digits of Table 252, the call are routed to Rerouting number.

Operation

1) Enable Co Call Rerouting ON and Save
2) T-net FoPstn Table(333) is reduced and created new Col Call Rerouting table.
3) If Compare CO Group is not assigned, All incoming calls are compared with Compare digits.
4) If Compare Co group is 1 and incoming call is co group 2, it doesn't work.
5) If Compare digits are 454**, called number should be 454xx(xx is any two digits).
6) Item [CO + Rerouting Number] : if co access code is '0' and tel-number is 1234567, 01234567
7) Item [CO + Rerouting Number] : if individual co numbering code is 88 , co is 5 and tel-number is 1234567, 880051234567
8) Item [CO + Rerouting Number] : if co group numbering code is 8 , co group is 3 and tel-number is 1234567, 8031234567
9) If you want to reroute Net number, Rerouting Type should be set NET Type.
10) If you want to reroute internal station, Rerouting Type should be set DISA Type.
11) Example)

Index	Incoming CO group	Compare Code	CO Code + Telnumber	Routing Type
0	1	454	88005123456	N/A
1	2	456**	801123456	N/A
2	1	42*555	0123456	N/A
3	5	353	801123456	NET Type
4	5	401		DISA Type

Index 0) If Incomming CO digits are matched digit “454” and CO group 1, seize CO 5 and sned digit 123456
Index 1) If it’s matched digit “456**” and CO group 2, seize CO group 1 and sned digit 123456
Index 2) If it’s matched digit “42*555” and CO group 1, seize First Co and sned digit 123456
Index 3) If it’s matched digit “353” and CO group 5, seize CO group 1 and sned digit 123456 in case of transit out of Networking call.
Index 4) If it’s matched digit “401” and CO group 5, work as DISA.

Conditions

1) Caution : If Co call rerouting table is used, Fop table(333) is reduced. And so, if you don’t like to do it, you shoud not choice 252 table.
2) Tnet Fop table(333) and Rerouing table(252) is reduced as following by system capacity.

	Fop table(333)	Rerouting table(252)
50A/B, 100	: 100 → 20	→ 80
300	: 200 → 30	→ 170
600	: 300 → 50	→ 250
1200	: 600 → 100	→ 500

Programming

SYSTEM CO Call Rerouting (PGM252-Web only)
Tnet Fop Table(333)

Related features

Hardware

2.82 OUTGOING MAILBOX DESTINATION

Description

If a CO/IP incoming, Caller dials "0" when listening to a station’s VSF mailbox greeting the call is routed according to the "Outgoing Mailbox Destination" as programmed for the station.

Operation

VSF outgoing mailbox destination call will be routed automatically.

Conditions

1) When a set a unavailable destination as a outgoing mailbox destination, Call will be routed to the Attandant.

Programming

STATION STATION ATTRIBUTES (PGM114-BTN19)

Related features

Hardware

2.83 REMOTE CONTROL WITH MOBILE EXTENSION

Description

Mobile extension user can change the mobile extension settings remotely.

Operation

1. The mobile extension user dials his DID number with mobile phone. Then the system will check the CLID, answer the call and the user will receive internal dial tone
3. He can enter the “remote control menu” with the code (remote MEX control code : PGM109-Flex16) and then confirmation tone will be provided.
4. He can select the desirable menu
5. He hears the dial tone. (Step 2)

Remote control menu

-. 1: Activate Mobile extension

-. 2: Deactivate Mobile extension

-. 3: Fwd to VSF/VM group

-. 4: Cancel Fwd to VSF/VM group

-. 5: ACD Agent Duty OFF

-. 6: ACD Agent Duty ON

Conditions

1. In menu 1(Activate Mobile extension), PGM Authority(PGM236-F1) should Enable
2. In menu 3(Fwd to VSF/VM group),

   VSF/VM hunt group(PGM190) should be existed.

   Call fwd admin (PGM111-F2) should be ON.

   VSF Access admin(PGM113-F2) should be Enable
3. In menu 5(ACD Agent Duty OFF), the reason of Duty OFF is 1.
4. Remote MEX Control code(PGM109-F16) should not be started with ‘#’.

Programming

NUMBERING PLAN 1. Remote MEX Control (PGM109-Btn 16)

Related features

Hardware

2.84 Prefix Dialing Table

Description

PGM 206 – Prefix Dialing Table. With this table, three features can be supported.

1. Analog CO Call Charge with NPR metering.
2. SIP Direct dialing with no wait inter-digit timer.
3. ISDN Prefix Call – ISDN enblock Dialing with Prefix Call Setup.

If first some digits(up to 8 digits) of outgoing dial number are matched with Prefix Code of each table, this table can start work. By each Co-line(PGM 142 – F20), Table ID(0-6) can be set. This table ID(PGM 142 – F20) is associated with PGM 206 – each table ID.

Operation

1. Analog CO Call Charge with NPR metering

1. Outgoing Call is established with analog Co-line.
2. Outgoing digits are matched with prefix code.
3. If Call Charge Timer is set to proper value(ex : 30 sec), call metering pulse is calculated one time per this timer.
4. For example, If Call charge timer is 30 sec, Call duration is 5 minute, and SMDR Cost Per M etering Pulse is 1.5$, then Call Cost is [1.5 * 10 = 15$].

SIP Direct dialing with no wait inter-digit timer

1. Outgoing Call is established with SIP Co-line.
2. Outgoing digits are matched with prefix code.
3. If total dialing digits are same as MAX DIGIT, then SIP outgoing call established directly.
4. So, user need not wait until inter-digit timer expiring.

ISDN Prefix Call – ISDN enblock Dialing with Prefix Call Setup in Russia

1. Outgoing Call is established with ISDN Co-line.
2. Outgoing digits are matched with prefix code.
3. If total dialing digits are same as MAX DIGIT, then ISDN outgoing call is established directly. So, user need not wait until inter-digit timer expiring.
4. If total dialing digits are less than MAX DIGIT, then ISDN outgoing call is established after int er digit timer expiring.
5. If total dialing digits are less than MIN DIGIT, then ISDN outgoing call is not established altho ugh prefix code is matched.
6. When ISDN setup is established, Num Of Type, Numbering Plan, Sending Complete option i s applied.

Conditions

1. If outgoing dialing digits are not matched with prefix code, this feature is not working.
2. Table ID is used commonly at three feature.
3. In case of Analog Call Charge, related program fields are Prefix Code, Call Charge Type, Call Charge time.
4. In case of SIP direct outgoing, related program fields are Prefix Code, Max Digit.
5. In case of ISDN enblock dialing, related program fields are Prefix Code, Min Digit, Max Digit, Num Of Type, Num Plan, Sending Complete.

Programming

1. PGM 206 – Prefix Dialing Table.
2. PGM 142 – Prefix Table ID.

Related features

Hardware

2.85 IP CALL RECORDING

Description

System can record automatically or manually using IPCR server. IPCR(IP call recording) Server can be registered to P5.5 system. The station with agent ID is automatically recorded about call , external and conference call.

Operation

Registration

Before registration, you should install the IPCR server in PC based on linux(os:fedora 12) using install CD or downloading from our bcs web site.

1. IPCR setting before registration to system.
   1.1) PBX registration(system IP, SIP ID, SIP Password)
   1.2) IPCR Server registration
   1.3) User registration
   1.4) Channel registration
2. System should set register MAC table for IPCR's MAC or Dip Switch ON of Register.
3. If SIP ID is not allowed, it's set SIP ID and password in PGM 443.

Programming Agent ID

1. Enter the number of IPCR's order in PGM 237.
2. Match Agent ID to favorite station.
3. You can see the ACR(Auto-call recording) or ODR(On Demand Recording).

4. You should choice STN Type for station. But DID Type is for the future.

Two way recording

1. You should set External VM group(ex: 620) and the SIP number of the IPCR as the member of Ex.VM in PGM190.
2. Automatic Recording Destination should be set the Ex.VM(ex: 620) of IPCR in station PGM(111~113). And you don't need to set automatic call recording ON because it's set automatic ally if you assigned the Agent ID to the station. You can set automatic call recording ALL or CO. In case CO, it's recorded only CO.

Operation

1. IP-Phone(S100) without agent ID answered from IP-phone(S101) with agent ID(A500).
2. If S101 can conference 3 way, S101 connects IPCR with the agent ID(A500).
3. If S101 cannot conference 3 way and there is MCIM, S100,S101 and IPCR are connected wi th MCIM for 3-way conference.
4. If you have flex button with two-way record of E.VM(620) and agent ID is ACR, it's flased during two way recording. But it's ODR, first time, it's not flashed. After pressing the flex button(twoway record), it's flashed. ACR is unconditionally recorded after connection and ODR is conditionally by user's choice.
5. Even though it's ODR, it can be recorded during talking. If user don't press the two-way recording button within talking, it's erased.

Conditions

1. You can search the recorded using Web Admin of IPCR.
2. IPCR server can be registered up to 10 servers in a system.

Programming

1. PGM 237 – Agent ID Table.
2. PGM 111~113 – Set Hunt group number(Ex.VM) of IPCR

59	☐	**Automatic Talk Recording Dest.**

3. PGM 190 - Set Hunt group number and member of IPCR

Related features

Hardware

2.86 3RD PARTY CALL RECORDING

Description

System can record automatically or manually using 3rd party call recording server. The Server can be registered to P5.5 system. The station with agent ID is automatically recorded about call , external and conference call.

Operation

Registration

1. SIP phone of 3rd party server can be registered.
    1.1) System need one more SIP unlock key.
    1.2) ID and password for SIP phone should be set in 3rd party server
    1.3) If SIP ID is not allowed, it's set SIP ID and password on PGM 443 in system.

Programming Agent ID and VM group

< VM group setting >

1. VM xxxx SERVER TYPE is 3RD TYPE in PGM 191-B10.
2. VM xxxx SERVER Number is Server number of PGM 237 in PGM 191-B11.
3. VM xxxx MEMBER TYPE is SIP TYPE in PGM 191-B12.
4. Match Agent ID to favorite station in PGM 237.

Two way recording

1. You should set External VM group(ex: 620) and the SIP number of the 3rd party server as the member of Ex.VM in PGM190.
2. Automatic Recording Destination should be set the Ex.VM(ex: 620) of 3rd party server in station PGM(111~113). You can set automatic call recording ALL or CO. In case CO, it's recorded only CO.

Operation

1. It's followd as IPCR call recording way.
2. You can do manually recording by information from 3rd party sever using SIP information.

Conditions

1. 3rd party server can be registered up to 10 servers in a system.
2. This 3rd party call recording server has only the member of SIP type. You don't use it by SLT type.

Related features

Hardware

2.87 FORCED DISCONNECT BUTTON (EMERGENCY SUPERVISOR BUTTON)

Description

Each Digital station that have flexible button can make [Forced Disconnect] button.

[PGM] + [Flex button] + [PGM] + [7#] + [HOLD/SAVE].
By using this button, busy co-line can be seized forcedly, busy station can be intrude forcedly.

Operation

To activate forced co-line disconnect feature:

1. Seize CO line.
2. The user hears busy tone and press [Forced Disconnect] button.
3. Busy line goes to idle state forcedly and the user hears a dial tone.

To activate forced intrusion to a busy station feature:

1. Call to busy station.
2. The user hears busy tone and press [Forced Disconnect] button.
3. Busy station connected with the user forcedly.

Conditions

1. SLT user can not use this feature.
2. If the station was talking to other station or CO line, the call is disconnected when the station receives forced Intrusion call.
3. The busy station is connected to forced intrusion caller only after short indication with beep tone.
4. Forced Intrusion call is not possible to a station in DND state.
5. Forced Intrusion call is not possible to attendant.

Programming

Related features

Hardware

2.88 CALL BACK TO CO

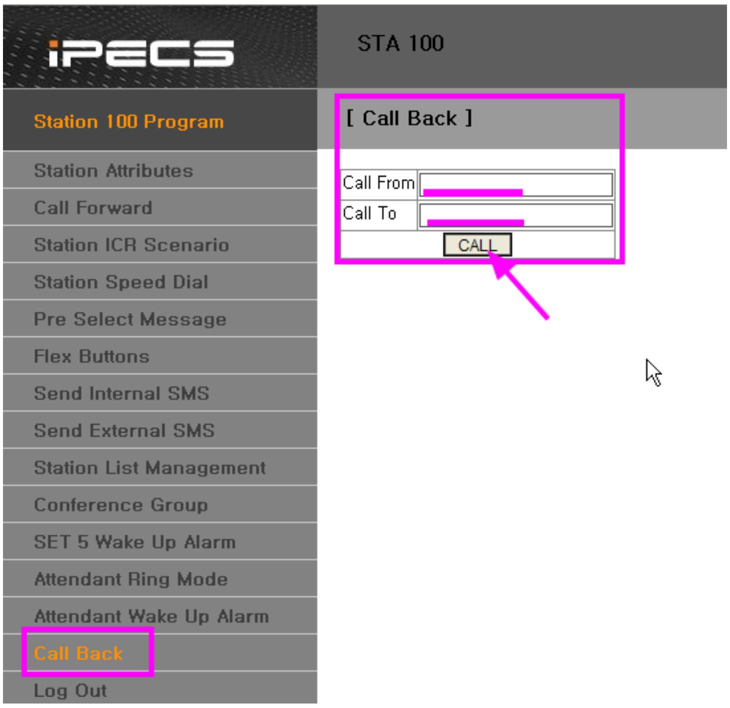
Description

User can call back to CO using station web window. Call From Number can call to Call To Number using own system resource. SMDR and COS is followed the station number.

Operation

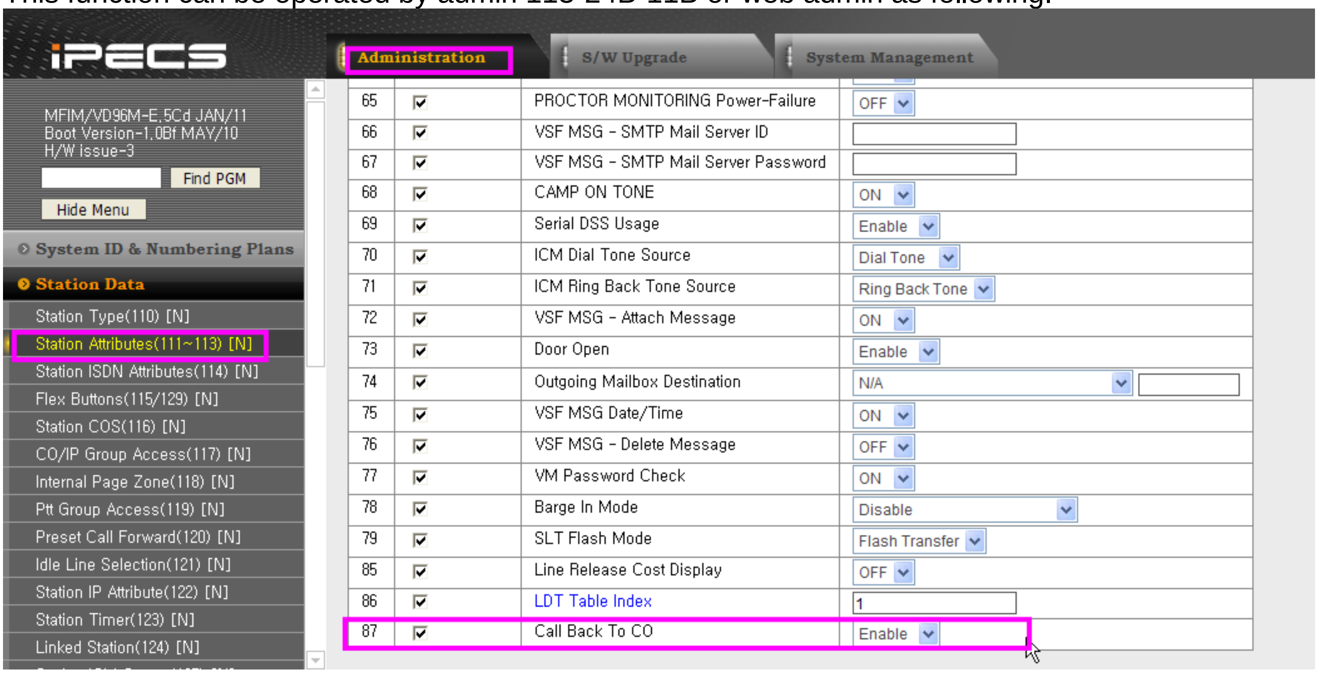
To activate make call back to CO:

1. The user can enter to station web using own station password.
2. The user should enter right digits to ‘From’ and ‘To’ box after that, you should press ‘CALL’button.
3. You can see the result that it’s success or not in widow display.



Conditions

1. This function can be operated by admin 113-24B-11B or web admin as following.



2. If Call Back To CO is disabled, you cannot operate it and you can see the error display.

Please Set Call Back Enable in Progamming

3. If it's not adaptable COS, you can see as following.

CO is not TOLL FREE

4. If there is no available Co, you can see the error message, 'There is no available CO [Call From Number]'.

5. If there is no adaptable Co, you can see the error message, 'There is no available Digital CO or LCO with LoopSupervisionTmr'.

Programming

1. Call Back To CO (PGM 113-Btn 24 – Btn 11)

Related features

Hardware

2.89 Personal Group

Description

Several stations can share same station number. That means Personal Group is extended feature of Linked Pair.
A Personal Group is composed with a master station and several member stations.
A master station and all member stations share master station's number.
By using this shared number, almost features (Call To xxx / Call From xxx / SMDR / Message Wait…) can be activated.
But, some features can be chosen by PGM 261 attribute about all station activating or individual station working.
In PGM 260, Personal group master and member can be assigned.
In PGM 261, Personal group attribute can be set.

Operation

Personal Group and Call Coverage
If master and members are assigned in admin 260, call coverage programs are set automatically as following.
In Master station attribute :
-. Call coverage(CC) usage : ON
-. CC on busy usage : ON
-. CC Mobile ext usage : ON
-. CC Delay Ring by members : ON
-. CC Service for wake up ring : ON
In Member stations attribute:
-. "Call for button" is assigned to max button in each members.
-. The initial value of delay is '0'.

Call to Personal Group.
Master number Display in caller
Members are ringing with delay of Call Coverage.
All leaved messages can be checked a Master station as well as member stations.

Call from Personal Group Master or Personal Group Member
1) With Master station number, all call can be made.
2) Each member can leave message by the master station number.

Conditions

Programming

PGM 260 – Personal Group Assign.
PGM 261 – Personal Group Attribute.

Attribute Name	Description
Use Master Wake Up	If this value is set to ON, all member wake-up follow by master wake up. If this value is set to OFF, individual wake-up is worked by each station.
Use Master Call Forward	If this value is set to ON, call forward setting affect to Master and all member. If this value is set to OFF, individual Call Forward is worked by each room.
Use Master DND	If this value is set to ON, DND setting affect to Master and all member. If this value is set to OFF, individual DND is worked by each room.

Related Features

SMDR
Call Coverage
Wake Up
Message Wait / VM Message

Hardware

3. INTERCOM

3.1 DIRECT STATION SELECT/BUSY LAMP FIELD (DSS/BLF)

Description

When a Flex button on an iPECS Phone or DSS Console is assigned as a **{DSS}** button, it also serves as a Busy Lamp Field. The LED indicates the status of the associated station or system facility.

Operation

iPECS Phone

Operation of this feature is automatic for assigned Flex buttons.

Conditions

1. A station receiving ICM ringing is considered busy and the associated station LED on all other stations will flash at 30 ipm.
2. A station receiving ICM ringing will receive visual indication by a flashing LED of the Flex button associated with the calling station.
3. When a station receives a Camp-On the LED of a DSS button associated with the calling station will flash.
4. The station is considered busy when:
   - in use,
   - receiving ICM Ring at an iPECS Phone,
   - receiving any ring at an SLT.

Programming

Related Features

Intercom Call (ICM Call)
Station User Programming & Codes

Hardware

3.2 INTERCOM CALL (ICM CALL)

Description
A non-blocking ICM is available to all stations in the system. Users may place an intercom call to other stations in the system by dialing applicable digits as defined in the system's Numbering Plan.

Operation
iPECS Phone

To place an intercom call
1. Lift the handset or press the **[SPEAKER]** button to receive ICM dial tone.
2. Dial station number or press **{DSS/BLF}** button.
3. For ring-back tone, await answer or For Intercom splash-tone, speak and await answer.

SLT

To place an intercom call
1. Lift the handset to receive ICM dial tone.
2. Dial station number.
3. For ring-back tone, await answer or,
   For Intercom splash-tone, speak and await answer.

Conditions
1. Intercom dial tone will time-out if action is not taken within Dial-Tone Time or if the time between digits exceeds the Inter-digit Timer. Error tone is received should a time-out occur.
2. ICM Dial tone is removed after dialing the first digit.
3. If the called station is busy, Intercom Busy tone is provided for the Busy Tone time (7 seconds) then, error tone is sent by the system. The caller may disconnect or activate a feature such as Message Wait/Callback prior to the time-out.
4. For iPECS Phone users, consecutive Intercom calls can be placed without the need to regain ICM dial tone (no need to hang-up) between calls; the user simply presses another **{DSS/BLF}** button.
5. An Intercom call to a station in the HF answerback or Voice Announce mode (H or P Intercom Signaling Mode) is not considered answered unless the called user lifts the handset or presses the **[SPEAKER]** button (goes off-hook).

Programming
STATION	1.	Prime Line (PGM 113-Btn 7)
SYSTEM	1.	ICM Dial Tone Timer (PGM 181-Btn 6)
	2.	Inter Digit Timer (PGM 181-Btn 7)

Related Features
Intercom Signaling Mode
Speakerphone

Hardware

3.3 INTERCOM CALL HOLD

Description

While on an active ICM Call, users of iPECS Phones can place the ICM Call on hold. The held station will receive the assigned Music-on-Hold. The call is placed on Exclusive Hold and recalls to the holding station after expiration of the Exclusive Hold Recall Timer.

Operation

iPECS Phone

To place an active ICM call on hold

1. Press the **[HOLD]** button, the **[ICM]** button LED will flash at the exclusive hold rate. ICM dial tone is received.

To retrieve the held ICM call

1. Press the **[ICM]** button or the **{DSS/BLF}** button associated with the held station, the **[ICM]** button LED illuminate and the ICM call connected.

Conditions

1. Only one ICM call may be placed on hold at a station.

Programming

Related Features

MOH (Music-On-Hold)
Intercom Call (ICM Call)
Exclusive Hold
Hold Recall

Hardware

iPECS Phone

3.4 INTERCOM CALLER CONTROLLED ICM SIGNALING

Description

A user can change the signaling mode of an Intercom call from Tone ring to Voice announce or Voice announce to Tone ring.

Operation

iPECS Phone

To change the ICM Signaling mode;

1. Place intercom call.
2. Dial '#', ICM Signaling mode will change from Voice announce to Tone ring or Tone ring to Voice announce.

SLT

To change the ICM Signaling mode;

1. Place intercom call as normal.
2. Dial '#', ICM Signaling mode will change from Voice announce to Tone ring or Tone ring to Voice announce.

Conditions

1. The ICM Signaling mode cannot be changed when an Intercom call is placed to a Linked Pair station.
2. If the signaling mode is changed, the call is not subject to Call Forward, No Answer.
3. The signaling mode for a specific Intercom call can only be changed once and cannot be changed back to the original signaling mode.
4. Changing the signaling mode does not affect privacy at the called station.

Programming

STATION 1. Caller Controlled ICM Signaling (PGM 111-Btn 15)

Related Features

Intercom Signaling Mode
Linked Station Pairs

Hardware

3.5 INTERCOM LOCK-OUT

Description

If the user takes no action after going off-hook for the Dial Tone timer or fails to dial an additional digit within the Inter-digit timer, the station will receive error tone for 30 seconds and be placed out-of-service (locked-out). The LED of associated **{DSS/BLF}** buttons as well as the station's **[ICM]** button flutter rapidly to indicate the out-of-service state.

For iPECS Phone users, if the **[SPEAKER]** is used, the station will receive error tone for 30 second and then automatically return to idle.

Operation

System

Operation of Intercom Lockout is automatic based on the Dial Tone & Inter-digit timers.

Conditions
1. If the station is assigned Howler Tone, error tone is presented for 30 seconds followed by 30 seconds of Howler tone followed by lockout and silence.

Programming
STATION
1. Howler Tone (PGM 111-Btn 5)

SYSTEM
1. ICM Dial Tone Timer (PGM 181-Btn 6)
2. Inter Digit Timer (PGM 181-Btn 7)

Related Features
Intercom Call (ICM Call)

Hardware

3.6 INTERCOM STEP CALL

Description
When busy tone is received on a dialed Intercom call, the user may place a call to another station by dialing the last digit of the station number. The system replaces the last digit of the previously dialed busy station with the dialed digit and places an Intercom call to the new station number.

Operation
iPECS Phone

To activate step call, while receiving busy on a dialed Intercom call;
1. Dial a digit other than the last digit of the busy station's intercom number.

Conditions
1. If the user dials the last digit of the busy station, Camp-On will be activated.
2. After receiving busy tone, if the user takes no action for the Busy Tone timer, 7 seconds, the system will start the Intercom Lockout procedure.

Programming

Related Features
Intercom Lock-Out
Intercom Call (ICM Call)

Hardware

3.7 Intercom Tenancy Group

Description

Stations in the iPECS are assigned to an Intercom Tenancy Group, group 00 ~ 15. Stations in a group are allowed or denied the ability to place intercom calls to stations in other groups on a group-by-group basis.

Each Intercom Tenancy Group is assigned an Attendant station. All dial '0' calls from a station in the group are routed to the assigned Attendant. In addition, the assigned Attendant can control the Day/Night Ring mode for stations in the group switching from Day to Night mode. Each group is assigned a separate Auto Ring Mode Table to change the Ring and COS mode automatically during the day and night service mode. In addition, DID calls to the system can be routed to a specified Intercom Tenancy group. By defining the group as the destination in the Flexible DID Conversion table calls will follow the Tenancy Group Auto Ring Mode table.

Operation

System

Operation of Intercom Tenancy Groups is automatic when programmed.

Conditions

1. Intercom calls from a station to a denied access Intercom Tenancy Group will return error tone.
2. Intercom Tenancy does not affect the Station Numbering Plan in the system. All stations in the system must have different station numbers even if they are assigned to different Intercom Tenancy groups.
3. The Attendant of an ICM Tenancy Group can be any station in the system and it is not affected by ICM Tenancy Group access.
4. When the Attendant of an ICM Tenancy Group sets Day/Night/Weekend mode, it will affect only the assigned ICM Tenancy Group.
5. Calls to and from CO/IP Lines are not affect by ICM Tenancy however; calls can not be transferred between groups if access is not allowed between the groups.
6. Intercom Tenancy Group 00 is the default or unassigned group. Stations assigned to group 00 are unaffected by Intercom Tenancy and can place and receive calls with stations of all other groups.

Programming

STATION

1. ICM Group Number (PGM 111-Btn 17)
2. ICM Tenancy Group (PGM 125)

TABLES

1. Flexible DID Table (PGM 231)
2. Auto Ring Mode (PGM 233)

Related Features

Intercom Call (ICM Call)
CO/IP Ring Assignment
Flexible Numbering Plan
Call Transfer
Auto Service Mode Control

Hardware

3.8 INTERCOM TRANSFER

Description

Users can transfer an active Intercom call to other stations in the iPECS system. Intercom calls can be transferred after announcing the call (screened) or without announcing the call (unscreened).

The Intercom station is placed on Exclusive Hold. The Transfer Recall Timer is initiated and, if this timer expires before the Intercom call is answered, the call will recall the transferring station until answered or abandoned.

Operation

iPECS Phone

To perform an Screened ICM transfer, while on an ICM call

1. Press **[TRANS]** button.
2. Dial the station to receive call.
3. At answer or Splash tone, announce call.
4. Hang-up, return to idle.

Or,

1. Press **{DSS/BLF}** button for the desired station.
2. At answer or Splash tone, announce call.
3. Hang-up, return to idle.

While on a Intercom call, Unscreened call transfer

1. Press **[TRANS]** button.
2. Dial Station to receive call.
3. Hang-up, return to idle.

Or,

1. Press **{DSS/BLF}** button for the desired station.
2. Hang-up, return to idle.

SLT

To perform a Screened transfer of an active Intercom call

1. Momentarily depress the Hook-switch.
2. Dial Station to receive call.
3. At answer or Splash tone, announce call.
4. Hang-up, return to idle.

While on an Intercom call, Unscreened call transfer

1. Momentarily depress the Hook-switch.
2. Dial Station to receive call.
3. Hang-up, return to idle.

Conditions

1. The **[ICM]** button provides an appearance for the transferred station. The LED indicates status and pressing the button connects to the station.
2. If the receiving station is busy, the transferring station may camp the call on to the busy station, see Camp-On.
3. A station in DND or out-of-service cannot receive a transfer, and such attempts will result in error tone.

Programming

Related Features

Exclusive Hold
Hold Recall
Call Waiting/Camp-On
DND (Do Not Disturb)
Call Transfer

Hardware

3.9 MESSAGE WAIT/CALL BACK

3.9.1 CLI Message Wait

Description

When an iPECS Phone with display receives an external call with Calling Line Identification and the call is abandoned (disconnects before answer), the system will generate a call log with the Caller Identification, date and time. The user may employ this log to review and, if desired, return the call.

An iPECS Phone user may assign a **{CLI MESSAGE WAIT}** button with direct access to the CLI Message Wait list. The button LED flashes at 30 ipm to indicate a CLI Message Waiting.

Operation

iPECS Phone

*To assign a **{CLI MESSAGE WAIT}** button*

[PGM] + **{FLEX}** + **[PGM]** + 32 + **[SAVE]**

To review CLI Messages

1. Press the **{CLI MESSAGE WAIT}** button, the first CLI message is displayed.
2. Press the **[VOL UP]/[VOL DOWN]** button to scroll through the CLI Messages.

```
165.123.23.36
21/11 09:10 CNT: 2
```

To delete the current CLI Message

1. Press the '*' button, the next CLI message is displayed.

To delete all CLI Messages

1. Press the **[SPEED]** button.
2. Press the '*' button twice.

To return a call to the current CLI Message

1. Press the **[SAVE]** button.

Conditions

1. The total number of Message Wait indications in queue for the system is 1000. Should the message Wait buffer fill, the buffer full message, as below will be output over the RS-232 port.
   "Warning: Message Wait Buffer Full."
2. The CLI Message Wait display includes the date, time, message count and CLI. The count is the number of attempts by a specific calling party with the time and date from the most recent attempt.
3. The CLI Message Wait status is stored in non-volatile memory to protect against power loss.

4. When a CO/IP call is assigned to ring multiple stations, or a group, CLI Message Wait is not provided.
5. When a CO/IP call with CLI is routed to an ACD/Terminal/Circular station group, the CLI Message Wait will be provided to the last station rung before the call was abandoned.
6. Once a call is answered by a station in the system, CLI Message Wait is no longer available i.e.; transferred calls do not provide CLI Message Wait.
7. A CLI Message Wait indication is left at the originally called station even if the call is forwarded to another station.
8. While idle, the user may toggle between CLI number and CLI name, if available, by pressing the **[TRANS]** button.
9. The CLI and called station number will be shown in the SMDR record.
10. For ISDN calls, the CLI Message Wait will display and dial the number as shown in the Table below, modified based on assignments in **PGM** 200 as follows:

    Incoming Prefix Code
    My Area Code
    My Prefix Area Code
    International Access Code

Inc Prefix	Area Code	Country Code	Access Code	Received Caller ID	Display
No	0343	82	00	National: 0343 50-7951	507951
No	0343	82	00	National: 0343 50-7951	507951
No	0345	82	00	National: 0343 50-7951	0343 507951
Yes	0343	82	00	National: 0343 50-7951	507951
Yes	0343	82	00	Subscriber: 0343 50-7951	0343 507951
Yes	0343	82	00	International: 492 888 7777	00492 888 7777

Programming

STATION
1. CLIP LCD Display (PGM 114-Btn 1)
2. CLIP LCD MSG WAIT EN/DIS (PGM 114-Btn 10)

SYSTEM
1. Incoming Prefix Code Insertion (PGM 143-Btn 12)
2. International Access Code (PGM 143-Btn 14)
3. My Area Code (PGM 143-Btn 15)
4. My Prefix Area Code (PGM 143-Btn 16)

Related Features

Station Message Wait/Call Back
ISDN (Integrated Service Digital Network)

Hardware

iPECS Phone

3.9.2 Short Message Service (SMS)

Description

The Short Message Service (SMS) provides the ability to send and receive text messages to and from iPECS phones equipped with a display including the Phontage, UCS Client and WLAN Phone. The text can comprise words, numbers, or an alphanumeric combination. Each short message can be up to 100 characters in length when Latin alphabets are used.

Users send and receive messages from their iPECS Phone or the Station Program selection in the iPECS system Web Home Page. For operation using the iPECS system Home Page see your System Administrator.

Operation

To assign a Flex button as a ***{SMS SEND}*** *button;*

[PGM] + **{FLEX}** + **[PGM]** + '36' + **[SAVE]**

To resend an existing SMS message;

1. Press the **[PGM]** button.
2. Dial '36', the SMS send code.

SMS SENDING MODE RESEND-(1) EDIT-(2)

3. Dial '1' to resend a message.

DIAL STA DEST(XXX-XXX) SKIP-(*)

4. Dial the desired station range to receive the message or '*' to use the displayed station range.

STA 100-124 SEND-(SAVE) CANCEL-(#)

5. Press the **[SAVE]** button to send the message or "#" to cancel.

To send a new message

1. Press the **[PGM]** button.
2. Dial '36', the SMS send code.

SMS SENDING MODE RESEND-(1) EDIT-(2)

3. Dial '2' to send a new message.

DIAL STA DEST(XXX-XXX)

4. Dial the desired station range to receive the message

STA 100-124 SEND-(SAVE) CANCEL_(#)

5. Dial the message using two digits for each character as shown in the table below, use the **[REDIAL]** button to delete a character.

Q – 11 Z – 12 . – 13 1 – 10	A - 21 B - 22 C - 23 2 - 20	D - 31 E - 32 F - 33 3 - 30
G – 41 H - 42 I - 43 4 - 40	J - 51 K - 52 L - 53 5 - 50	M - 61 N - 62 O - 63 6 - 60
P - 71 R - 72 S - 73 Q - 7* 7 - 70	T - 81 U - 82 V - 83 8 - 80	W - 91 X - 92 Y - 93 Z - 9# 9 - 90
*1 - Blank *2 - : *3 - ,	0-00	#

6. Press the **[SAVE]** button to send and store the message or '#' to cancel.

To retrieve SMS Messages:

1. Press **[MSG/CALLBK]** button. The message contents summary will be shown as below.

ST	CL	VS	VM	FS	MS
001	001	005	006	001	004

2. Dial '6' to select MS, SMS message wait.
   '1' ➔ ST: Station Message wait,
   '2' ➔ CL: CLI Message wait
   '3' ➔ VS: VMIM/VSF message wait
   '4' ➔ VM: external Voice Mail
   '5' ➔ FS: Feature server
   '6' ➔ MS: SMS message wait.

The display shows the first line of received SMS messages.

MSG: [0] EMERGENCY CONFER [1] LETS GO TO LUNCH

3. Use the **[VOL UP]**/**[VOL DOWN]** to scroll through messages.

MSG: [2] WE HAVE A MEETING A [3] CONFERENCE AT 9 AM

4. Dial the message number ('01'~'10') to view the entire message, use the **[VOL UP]/[VOL DOWN]** to view each line of the message.

MSG:> WE HAVE A MEETING AT >7PM IF YOU CAN NOT MAKE IT

To delete SMS message

1. Dial '#'.

DELETE MESSAGE? YES : 1 NO : 2 ALL:3

2. Dial '1' to delete current message, '2' to return to idle or '3' to delete all received SMS messages.

Conditions

1. The system can store up to 10 received SMS messages and 1 sent SMS message per station.
2. Once the received message storage capacity is reached, additional incoming SMS messages are ignored.
3. For operation with the Phontage, WLAN Phone or UCS Client, refer to the respective User Guide.

Programming

Related Features

Hardware

iPECS Phone w/Display
iPECS Phontage
iPECS WLAN Phone
iPECS UCS Client

3.9.3 Station Message Wait/Call Back

Description

When a called station does not answer or is in DND, a station user can activate a Message Wait indication to request a Call Back. A station may receive a Message Wait from any number of other stations in the system. The station receiving the Message Wait can return the calls using the **[MSG/CALLBK]** button.

When a busy station is called, the calling user may request to be placed in a queue to receive a Call Back. When the called station returns to idle, the system signals the initiating station with Callback ring. When the user answers, the previously busy station is called.

When a Message is waiting, the **[MSG/CALLBK]** button LED of iPECS Phones and digital phones flashes and, for properly equipped SLTs, the MW Lamp will flash.

Operation

iPECS Phone

To leave a Message Wait, while receiving ring back tone or no response on a call announce (H or P mode)

1. Press the **[MSG/CALLBK]** button, confirmation tone received.
2. Hang up, Message Wait activated.

To leave a Message Wait, while receiving DND tone

1. Press the **[MSG/CALLBK]** button, confirmation tone received.
2. Hang-up, Message Wait activated.

To leave a Call Back (queue for a station), while receiving busy

1. Press the **[MSG/CALLBK]** button, the user receives confirmation tone.
2. Hang up, return to idle.

To respond to a Call back recall, when the busy station is available the system calls back,

1. Lift the handset or press the **[SPEAKER]** button.
2. Previously busy station is called.

To retrieve Station Messages Waiting

1. Press **[MSG/CALLBK]** button.

Either the message contents summary will be shown as below or the Station Messages Waiting will be listed, see step 3 below.

ST	CL	VS	VM	FS	MS
001	001	005	006	001	004

2. Dial ‘1’ to select ST (Station Message Wait)
   - ‘1’ → ST: Station Message Wait,
   - ‘2’ → CL: CLI Message Wait
   - ‘3’ → VS: VMIM/VSF Message Wait
   - ‘4’ → VM: commercial Voice Mail
   - ‘5’ → FS: Feature server
   - ‘6’ → MS: SMS message wait.
3. Press the **[VOL UP]/[VOL DOWN]** button to scroll through the Station Messages

To return a call for the current Station Message

1. Press the **[SAVE]** button.

To delete the first Message Wait from the list

1. Press '*' button, the list is updated removing the first station number in the list.

To delete all Message Waits

1. Press **[SPEED]** button.
2. Press '*' button twice.

SLT

To leave a Message Wait, while receiving ring back tone or no response on a call announce (H or P mode)

1. Momentarily press the hook switch.
2. Dial '556', the Message Wait/Call Back code.
3. Hang up, Message Wait activated.

To leave a Message Wait, while receiving DND tone

1. Momentarily press the hook switch.
2. Dial '556', the Message Wait/Call Back code.
3. Hang up, Message Wait activated.

To retrieve a Station Message Wait;

1. Dial '557', the Message Wait/Call Back Answer code.

To leave a Call Back (queue for a station), while receiving busy

1. Momentarily press the hook switch.
2. Dial '556', the Message Wait/Call Back code.
3. Hang up, return to idle.

To respond to a Call back recall, when the busy station is available the system calls back,

1. Lift the handset.
2. Previously busy station is called.

Conditions

1. A Message Wait/Call Back return call will always ring at the receiving station overriding the Intercom signaling mode selected.
2. A station can leave only one callback request.
3. The **[MSG/CALLBK]** button LED will continue to flash until all Message Wait and Call Back requests, including Voice Mails, have been serviced.
4. If a station is attempting to leave a message and the system Message Wait queue is full, the station will receive ICM busy tone.
5. A Message Wait remainder tone can be enabled to remind the user of messages waiting.
6. A station in Call Forward can leave a message wait.

7. A Message Wait indication is left at the originally called station even if the call is forwarded.
8. An iPECS Phone with LCD may call back to stations that left messages in any desired order, or the normal (“oldest first”) order.
9. Placing an Intercom call to a station will cancel any existing Message Wait from that station.

Programming

SYSTEM 1. Message Reminder Tone Timer (PGM 181-Btn 8)

Related Features

Message Wait Reminder Tone
CLI Message Wait

Hardware

3.9.4 Message Wait Reminder Tone

Description

In addition to the **[MSG/CALLBK]** button LED, the iPECS Phones can be sent a tone as a periodic reminder to the user of message waits in queue. This tone is sent to the station only while idle and is heard over the speaker.

Operation

System

Reminder tone is sent to stations automatically when assigned.

Conditions

1. The interval between tones can be 00 to 60 minutes. A setting of 00 disables the reminder tone.
2. The reminder tone will continue until all messages have been retrieved.
3. A busy station or station in DND will not receive the Message Wait Reminder tone.

Programming

SYSTEM 1. Message Reminder Tone Timer (PGM 181-Btn 8)

Related Features

Message Wait/Call Back

Hardware

iPECS Phone

3.10 PAGING

3.10.1 Internal/External & All Call Page

Description

A station, which is permitted to access page facilities, can connect and transmit voice announcements to any or all of the systems Internal/External Page zones. Stations are grouped into "zones" to receive pages to the zone. Stations not assigned to any zone will not receive a page including All Call pages.
A page warning tone, if assigned, will be provided to the Page Zone(s) prior to the audio connection. The user is allowed to continue the page for the specified Page Time-out timer after which the user is disconnected and the Page Zone(s) is returned to idle.

The default Page Zone dial codes are as follows:

	iPECS-50	iPECS-100	iPECS 300 iPECS 600	iPECS 1200
Internal Page Zones	*101~*110	*101~*110	*101~*135	*101~*200
Internal All Call Page	#3	#3	#3	#3
External Page Zones	#41	#41~#46	#41~#46	#41~#46
External All Call Page	#5	#5	#5	#5
All Call Page	#00	#00	#00	#00

iPECS-Micro have same default Page Zone dial codes with iPECS-50 except External Page. (iPECS-Micro does not have external port.)

Flexible buttons of an iPECS Phone may be assigned to access a Page Zone as a **{PAGE ZONE}** button.

Operation

iPECS Phone

*To assign a Flex button as a **{PAGE ZONE}** button*

[PGM] + **{FLEX}** + Page Zone number + **[SAVE]**

To make a page,

1. Lift the handset.
2. Dial the desired paging code or press a **{PAGE ZONE}** button.
3. If assigned, after the Page Warning Tone, make announcement.
4. Replace the handset and go on-hook.

To queue for a page when busy is received.

1. Press the **[MSG/CALLBK]** button.
2. Replace the handset returning to idle.

SLT

To make a page,

1. Lift the handset.
2. Dial the desired paging code.
3. If assigned, after the Page Warning Tone, make announcement.
4. Replace the handset, go on-hook.

To queue for a page when busy tone is received.

1. Dial ‘556’, the Call Back code.
2. Replace the handset returning to idle.

Conditions

1. Stations, which are denied access to paging, will receive error tone when any Page Access code is dialed.
2. Stations dialing a Page Code will be queued when any of the other Internal or External Page zones are busy.
3. DTMF signals from stations are transmitted through the systems External Page port after access.
4. If an iPECS Phone user attempts to page using the speakerphone, pre-selection will be activated and the display shows "LIFT THE HANDSET TO PAGE".
5. Stations in DND or busy will not receive Page announcements.
6. Stations, which are not included in any Internal Page Zone, will not receive any page, including All Call.
7. For external paging, an external amplifier and speaker(s) are required.
8. The systems External Control Contacts may be assigned to activate when External Page is accessed.
9. A station is permitted only one Page Zone queue request at a time. If a station attempts another Page Zone queue, only the later queue request is honored.
10. When a busy Page Zone becomes idle, the system will select the oldest page queue, and signal the appropriate station. The signaled station will have an audible ring indicating the queue callback. The audible ringing will be a distinctive signal. The All Page Zone, while signaling a queued station, is considered busy. In addition, the All Page Zone is considered busy when any page zone is active.
11. The queue recall is always in the tone ring mode regardless of the station’s ICM signaling mode.
12. If the waiting station is idle, the Call Back ring signals the station for 15 seconds, after which, the queue is canceled and the next station in the queue is signaled.
13. If the waiting station is busy, and the Page zone becomes available the next idle station in the Page Queue list is signaled and the busy waiting station is placed at the bottom of Page Queue list. If there is no idle next station in the Page Queue, the Page Queue is canceled.
14. When the waiting station goes to idle, and both a “Page Queue” and “CO Call back Queue” exist, the Page Queue is given priority.

Programming

STATION	1. Page Access (PGM 111-Btn 7) 2. Internal Page Zone Access (PGM 118)
SYSTEM	1. Page Warning Tone (PGM 161-Btn 4) 2. External Control Contacts (PGM 168) 3. Paging Timeout Timer (PGM 181-Btn 9)

Related Features

Meet Me Page Answer

Hardware

External Amplifier & Speakers

3.10.2 Meet Me Page Answer

Description

Any station may respond to a "Meet Me" Page request over an Internal or External Page Zone. The user answers the page from any station and is connected to the paging party.

Flexible buttons of an iPECS Phone may be assigned as a **{MEET ME}** button.

Operation

iPECS Phone

To assign a Flex button as a {MEET ME} button

[PGM] + **{FLEX}** + '544' + **[SAVE]**

To answer a page with Meet Me Page

1. Lift the handset or press the **[SPEAKER]** button.
2. Dial '544', the Meet Me Page code or press the **{MEET-ME}** button.

SLT

To answer a page with Meet Me Page

1. Lift the handset to receive intercom dial tone.
2. Dial '544', the Meet Me Page code.

Conditions

1. A Meet Me Page must be answered within the Page Time-out timer.
2. A station may answer a Meet Me Page from any station regardless of pickup/paging group assignments and page access permission.

3. The paging party must remain off-hook until the paged party answers the Meet Me request. The initiator may press the Mute button to eliminate transmitting over the page circuit while waiting for the party to answer.

Programming

SYSTEM 1. Paging Timeout Timer (PGM 181-Btn 9)

Related Features

Internal/External & All Call Page

Hardware

3.10.3 SOS Paging

Description

The system allows of being recorded multiple VMIM messages for pre-recorded paging. Depending on circumstance, user can use pre-recorded messages for paging. Recorded VMIM message is paged to a page zone at emergency.

Operation

To assign **{VMIM SOS Paging}** at a flexible button, perform the following Steps:

1. Press the **[TRANS/PGM]** button.
2. Press the FLEX button to assign.
3. Dial paging code (5xx).
4. Dial Message number (001-070).
5. Press the **[HOLD/SAVE]** button.

To activate VMIM SOS paging:

1. Press the assigned **{VMIM SOS Paging}** flexible button.

Condition

- This feature can be only activated by pressing assigned flexible button in idle state.
- The VMIM message for SOS Paging can be recorded only at attendant station.
- Paging zone includes internal, external and all call paging area.
- VMIM SOS paging is not restricted by VMIM Paging timer. The whole VMIM SOS paging can be paged even though Paging Timeout timer expires.

Programming

STATION 1. Page Access (PGM 111-Btn 7)

Related Features

VMIM/VSF, Paging

Hardware

VSF and/or VMIM

3.11 PUSH-TO-TALK PAGING

Description

Each iPECS Phone can be assigned as a member of one or more of the system's nine Push-To-Talk page groups. The iPECS Phone user may login or log-out of any one or all PTT groups to which it is assigned. Once logged in, the user may place or receive one-way page announcements to/from other users who are logged in to the same PTT group. To place a PTT page announcement, the user must press and hold the **{PTT}** Flex button.

An Attendant may log other stations in and out of PTT groups.

Operation

iPECS Phone

*To assign a **{PTT}** Flex button;*

[PGM] + **{FLEX}** + **[PGM]** + '99' + **[SAVE]**

To log-in to a PTT group;

1. Dial '#0', the PTT Log-in/out code.
2. Dial the desired PTT group number ('1'~'9' and '0' for all groups).

To log-out of the PTT group(s);

1. Dial '#0', the PTT Log-in/out code.
2. Dial '*'.

To place a page to the active PTT group;

3. Press and hold the **{PTT}** Flex button.
4. After confirmation tone, make page announcement.

Attendant

To log other stations in to a PTT group;

1. Press the **[PGM]** button.
2. Dial '077', the Attendant PTT log-in/out code.
3. Dial the desired station range.
4. Dial the PTT group number ('1'~'9' and '0' for all groups).
5. Press the **[SAVE]** button.

To log other stations out of a PTT group;

1. Press the **[PGM]** button.
2. Dial '077', the Attendant PTT log-in/out code.
3. Dial the desired station range.
4. Dial '*'.
5. Press the **[SAVE]** button.

Conditions

6. Conditions associated with Internal/External & All Call Page apply to Push to Talk paging.
7. To access PTT paging, the station must be permitted access to system paging, **PGM CODE** 111, button 7.
8. If allowed access to all PTT groups, a station may log-into all groups (PTT group 0) to place announcements to all groups simultaneously and receive announcements from any group.
9. A station can only login to PTT groups to which it is assigned as a member.
10. The station must have a **{PTT}** to place or receive PTT announcements. As a default, the iPECS WLAN phone is assigned a PTT button.
11. The station may be assigned and logged in to the default active PTT group in the system database.

Programming

NUMBERING
1. PTT Log-in/out Code (PGM 109-Btn 4)

STATION
1. Page Access (PGM 111-Btn 7)
2. Default Active PTT Group (PGM 111-Btn 16)
3. PTT Group Access (PGM 119)

Related Features

Paging

Hardware

iPECS Phone
iPECS WLAN
iPECS Phontage
iPECS UCS Client

3.12 BARGE IN

Description

Barge in permits an authorized extension to intrude into other existing outside/internal calls.

Between intruding extension and parties on initial calls a conference call is established.
There are Two Barging In operations.
- Monitor : The intruding extension can listen to the existing conversation.
- Speech : The intruding extension can join to the existing conversation.
In PGM 113 – Barge In mode, Barge In supervisor mode can be set.
If 0, Barge In is disabled.
If 1, Barge In is possible only monitor other’s conversation.
If 2, Barge In is possible monitor other’s conversation, join it, and forced disconnected it.

Operation

Monitor:

1. Call to the busy station and user hears the busy tone.
2. Press the “MONITOR” button on 3 soft buttons. (Use Navi > Key to Display)
3. Caller can listen to the existing conversation. The others hears ‘Warning Tone’

Speech & Drop:

1. During caller listen to the existing conversation, press the “JOIN” button on 3 soft buttons.
2. Caller can join the conversation. The others hear “Intrusion Tone”.
3. Supervisor can cancel the conversation with “DROP” button on 3 soft buttons
4. Supervisor can exit from barge-in by hang up the phone.

Conditions

1. This feature is only supported for digital phone with 3 soft buttons.
2. This feature is only available for called party is talk state (with CO Line or Station).
3. Barge-In supervisor can see the other party of called station before monitor operation.
4. Barge-In to SIP extension is allowed only when MCIM is installed.

Programming

PGM 113 – Barge In Mode.

Related Features

Intrusion

Hardware

4. CO/IP

4.1 AUTO FAULT DETECTION AND RECOVERY

Description

If a CO line fault is reported from a PRI/VoIP/SIP gateway, the iPECS places the CO Line in an Out-Of-Service state and places it in the Unused CO/IP Group. Upon receiving the CO Line recovery report, the CO Line is automatically restored. The fault is also reported to the iPECS NMS when configured.

Operation

System

Operation of Fault Detection and Recovery is automatic.

Conditions

1. The "Unused" CO Group contains CO line numbers that are not used or are temporally blocked.

Programming

CO/IP 1. CO Line Group (PGM 141-Btn 1)

Related Features

CO/IP Line Groups

Hardware

4.2 CO LINE FLASH

Description

Analog CO lines recognize a brief open or ground connection, "Flash", as a request for new dial tone. When used behind a PBX, a Flash is often used to activate a PBX feature or call transfer.

Operation

iPECS Phone

While connected to an analog CO line

1. Press the **[FLASH]** button, the system generates a flash on the CO line.

SLT

While connected to an analog CO line

1. Momentarily depress the Hook-switch.
2. Dial '551', the Flash feature code.

Conditions

1. Stations may Flash on a CO Line defined for PABX operation and will experience COS dialing restrictions if a PABX Trunk access code is dialed.
2. During a Flash, the LED for the CO line button will remain lit.
3. A Flash may be stored as a part of a Station or System Speed Dial number.
4. While connected to an internal call or dial tone, pressing the **[FLASH]** button will regain ICM dial tone.
5. While making a Page Announcement (internal or external, including all call paging), pressing the **[FLASH]** button will terminate the Page Announcement and return ICM dial tone.
6. The Flash function is not available on VoIP calls.

Programming

CO/IP

1. Flash Type (PGM 141-Btn 6)
2. CO Flash Timer (PGM 142-Btn 12)

Related Features

Station Speed Dial
System Speed Dial

Hardware

4.3 CO/IP LINE GROUPS

Description
The CO/IP lines in the system can be placed together into groups for assigning access by stations and common access dial codes. There are 20 groups in the iPECS-Micro & 50 & 100 , 72 groups available in the iPECS 300 & 600 and 200 grouups available in the iPECS 1200. Unused CO lines are assigned to CO/IP Group 21 in the iPECS-Micro & 50 & 100 , 73 in the iPECS 300 & 600 and 201 in the iPECS 1200. Private Lines are assigned to CO/IP Group 00.

Operation

Conditions
1. Unused CO/IP lines should be assigned to the appropriate group so that stations cannot access these lines.
2. CO/IP lines in Groups 01 through 72(200) can be accessed individually by dialing the code '88' and the CO/IP line number.
3. Only {CO}/{IP} line buttons can access the CO/IP lines assigned to group 00, Private Lines.
4. The system will select a CO/IP line from a group based on the Round Robin or Last Choice determined by database assignments.

Programming
STATION 1. CO PGM (PGM 112-Btn 6)

CO/IP 1. CO Line Group (PGM 141-Btn 1)

SYSTEM 1. CO Line Choice (PGM 160-Btn 4)

Related Features

Hardware

4.4 CO/IP Line Preset Forward

Description

Each CO/IP Line can be assigned a Ring-No-Answer Preset Forward destination. An incoming call on the CO/IP line will be routed to the defined ring destination. At expiration of the CO/IP Line Preset No Answer Forward timer, the call is forward to the defined Preset Forward destination, which is an index to the ICLID Ring Assignment Table.

The destination can be a station or station group including an adjunct Voice Mail. When the call is forwarded to an adjunct Voice Mail group, a predefined Voice Mail Id (VMID) will be sent to the VM system to identify the Mailbox to receive the call.

Operation

System

Operation of this feature is automatic.

Conditions

1. CO/IP line Preset Forward is disabled for calls initially routed to a station group.
2. CO/IP line Preset Forward will override Call Forward No-Answer at a station.
3. CO/IP line Preset Forward is disabled if the Preset Forward Timer is set to 0.
4. The CO/IP line Preset Forward destination cannot be the VMIM/VSF group.

Programming

CO/IP	1. CO Preset Forward, Forward timer (PGM 147-Btn 1) 2. CO Preset Forward, Ring Table Index (PGM 147-Btn 2) 3. CO Preset Forward, VMID (PGM 147-Btn 3)
ISDN & ICLID	1. ICLID Ring Assignment Table (PGM 204)

Related Features

Call Forward
ICLID Call Routing

Hardware

4.5 CO/IP RING ASSIGNMENT

Description

Each station in the system can be programmed to provide an audible signal when the system detects an incoming call on specified CO/IP lines. Separate ring assignments are made for Day, Night and Timed Ring operation mode. In addition, the audible signal at the station can be delayed by 1 to 9 ring cycles allowing other stations to answer the call first.

Operation

System

Operation of this feature is automatic.

Conditions

1. Separate assignments are made for stations to ring in the Day, Night and/or Timed Ring mode.
2. Audible alerting for an incoming VoIP call is based only on the derived IP Address.
3. A busy station receives muted ring or Call Waiting tones as appropriate for the station's Off-hook ring assignment.
4. The system Ring mode can be selected manually or automatically. In the Automatic mode, Day/Night selection is determined based on the Automatic Ring Mode Selection table. The Attendant has manual control over the Ring mode selection.
5. The LCD of the Attendant station will display Night and Timed Ring Mode and the **[DND]** button LED will flash.
6. If a CO/IP line is not assigned to ring at any station, incoming calls on the CO/IP line will ring the first available Attendant.

Programming

CO/IP	1. CO Ring Assignment (PGM 144)
TABLES	1. Auto Ring Mode Selection Table (PGM 233)

Related Features

Day/Night/Timed/Scenario Ring Mode
Auto Service Mode Control
Off-Hook Signaling

Hardware

4.6 CO Line Release Guard Time

Description

To assure that the PSTN switching equipment has sufficient time to restore to the idle condition, the system will hold analog CO lines in a busy state to users after release of a CO line by a station. The time between the station disconnect and when the system changes the CO line status from busy to idle is the CO Line Release Guard time.

Operation

System

Operation of this feature is automatic.

Conditions

Programming

SYSTEM	1. CO Release Guard Timer (PGM 180-Btn 16)

Related Features

Hardware

4.7 CO Ring Detect

Description

The system incorporates timers for Ring-on and Ring-off durations to assure proper alerting. When the duration of the ring signal exceeds the Ring-on timer, alerting will start. When the ring is not present for a period exceeding the Ring-off timer, alerting will stop. This allows the system Ring cycle detection to be matched to the many and varied PBX systems.

Operation

System

Operation of Ring detect is automatic.

Conditions

1. Ring On and Ring Off are assigned on a system basis.
2. The CO Ring Detect is applied to analog CO Lines only.

Programming

SYSTEM	1. CO Ring OFF Timer (PGM 180-Btn 17)
	2. CO Ring ON Timer (PGM 180-Btn 18)

Related Features

Hardware

4.8 Dial Pulse Signaling

Description
An analog CO line will send dial pulse signals to the central office. If programmed as a pulse CO line, the system will send open loop pulses at 10 pps with the assigned break/make ratio.

Operation
System

Operation of this feature is automatic when programmed.

Conditions
1. The break/make ratio is system programmable as 60/40 or 66/33.

Programming
CO/IP
1. CO Line Signal Type (PGM 141-Btn 5)

SYSTEM
1. Pulse Dial Ratio (PGM 176)
2. Pause Timer (PGM 181-Btn 10)

Related Features
Dial Pulse to Tone Switchover
DTMF Signal Sending

Hardware

4.9 Direct Inward Dial (DID)

Description
A carrier service, known as Direct Inward Dial (DID), sends digits to the system so that the call may be routed directly to a specific station or system facility. Digits sent to the system are generally the last digits (3 or 4) dialed by the caller. DID service is available over analogue, digital, and packet networks. Analogue DID lines are limited to one-way incoming service and require special call-start signaling definition. ISDN lines can provide two-way, incoming DID and normal outgoing service, and require no special signaling.

After collecting the digits from the carrier, iPECS converts the digits employing one of three DID digit conversion Types:

- Type 0 - collect incoming digits based on the programmed digit count and convert the digits according to the DID conversion pattern, resulting in the DID destination number.
- Type 1 - use the incoming DID digits as the destination number without converting.
- Type 2 - Following the result of DID conversion Type 0, route the call according to the Flexible DID table conversion.

Operation

System

System response to an incoming DID call (analogue CO line):

1. set-up a connection based on the defined Start signal,
2. collect incoming digits based on the programmed Receive Digit Count,
3. handle digits based on the Conversion type (0-2),
4. route the call to assigned destination.

System response to an incoming DID call (ISDN line):

1. set-up a connection based on the received call set-up messages,
2. collect incoming digits and delete digits from left based on the programmed ISDN Remove Digit Count,
3. handle digits based on the Conversion type (0-2),
4. route the call to assigned destination.

Conditions

1. If ICLID routing is assigned for the CO/IP Line, the received Caller Id is compared to the ICLID Table for routing first. If Caller Id does not match an entry in the ICLID Table, the normal DID call processes are used.
2. DID calls that encounter a busy signal, are not answered in the DID/DISA No Answer Timer, or are received at a vacant or invalid number can be routed to the Attendant, a tone, Station group, or VMIM/VSF announcement. When the Attendant receives such calls, the call is appropriately identified by the Attendant iPECS Phone display.
3. For a station that is part of a non-pilot Station Hunt group, DID calls will follow the group hunt process if the Station is busy or does not answer the call.
4. DID calls are subject to Group Call Pick-up and Directed Call Pick-up.
5. If a VMIM/VSF announcement is defined as the destination in the Flexible DID Destination Table, a Caller Controlled Routing Table for the announcement can be defined. iPECS can be configured to drop (disconnect) the call after playing the recorded announcement.

Programming

STATION

1. SIP User ID Table (PGM 111-Btn 19)
2. Station SIP Attributes 2 (PGM 126-Web only)

BOARD

1. H323 VoIP Attributes
2. SIP Gateway Attributes

CO/IP
1. CO Service Type (PGM140)
2. ISDN DID Remove Digit Count (PGM 143-Btn 5)
2. DID Conversion Type (PGM 145)

SYSTEM
1. DID/DISA Busy Destination (PGM 167-Btn 1)
2. DID/DISA Error Destination (PGM 167-Btn 2)
3. DID/DISA No Answer Timer (PGM 181-Btn 2)

TABLES
1. CCR Audio Text Tables (PGM 228)
2. Flexible DID Table (PGM 231)

Related Features
VMIM/VSF-Auto Attendant
Directed Call Pick-Up
Group Call Pick-Up
IP Trunking
IP Address Dialing

Hardware
DID, BRI or PRI Module

4.10 DIRECT INWARD SYSTEM ACCESS (DISA)

Description
Each CO/IP path may be assigned for DISA service, which allows an incoming caller to gain access to the system resources and/or features. The iPECS will answer the outside call and provide the user intercom dial tone or route the call based on the VMIM/VSF Auto Attendant announcement settings where Caller Controlled Routing may be defined. The DISA caller may then access the desired resource using dial codes.

If an Authorization Code is required for DISA access, when the system answers an incoming DISA call, DND Warning tone is provided to indicate an Authorization Code must be entered.

Operation
System

Incoming call subject to DISA service
1. Recognize incoming call.
2. Answer call and connect caller to Intercom dial tone or AA announcement.
3. Process call based on received digits/programming.

DISA Caller

To access the system's resources from an external party

1. Place call to DISA facility of the system.
2. At receipt of Intercom dial tone/AA announcement, dial as desired. If DND Warning tone is received, enter an Authorization Code, to receive Intercom dial tone.

Conditions

1. Each CO/IP path is separately assigned for DISA operation during Day, Night and/or Timed system operation mode. DISA operation is active only when the system is in the assigned operating mode(s).
2. DISA callers can be routed to a VMIM/VSF Auto Attendant announcement in place of Intercom dial tone. The announcement can be associated with a CCR Table or assigned to disconnect after playback ('#').
3. A DISA caller can be required to enter an Authorization Code to access the system's external outgoing resources, facilities or features. If required, the caller is permitted to retry entry of a valid Authorization Code based on the DISA Retry count. Continued failure results in disconnect.
4. DISA callers are subject to COS dialing restrictions. If Authorization Codes are required and the code entered matches a Station Authorization Code, the station's COS will apply. Otherwise, the assigned DISA COS will apply. In both cases, the CO/IP COS for the outgoing path will be active.
5. The system will disconnect an outgoing DISA call if the Unsupervised Conference timer expires or disconnect supervision is received. A disconnect warning tone is provided 15 seconds prior to disconnect.
6. If a DISA caller encounters a system All Lines Busy, busy tone is received for 5 seconds before ICM Dial tone is again presented and the DISA caller may try another call.
7. LEDs associated with the DISA CO Line appearance will provide normal status indications at all stations except the Attendants. The LED for the line at an Attendant will flutter at 240 ipm when busy.
8. An iPECS Phone user can only receive a DISA call with an available CO/IP appearance button.

Programming

CO/IP	1. DISA Account Code (PGM 142-Btn 5) 2. DISA IP Access (PGM 142-Btn 11) 3. DISA Service Attributes (PGM 146)
SYSTEM	1. DISA Retry Counter (PGM 160-Btn 5) 2. DISA COS Assignment (PGM 166) 3. DID/DISA Busy Destination (PGM 167-Btn 1) 4. DID/DISA Error Destination (PGM 167-Btn 2) 5. DID/DISA No Answer Timer (PGM 181-Btn 2) 6. Unsupervised Conference Timer (PGM 182-Btn 5)
TABLES	1. CCR Audio Text Tables (PGM 228)

1. Auto Ring Mode Selection Table (PGM 233)

Related Features

VMIM/VSF-Auto Attendant
Day/Night/Timed/Scenario Ring Mode
Dialing Restrictions
Authorization Codes (Password)
Unsupervised Conference
VMIM/VSF Integrated Auto Attd/Voice Mail
Auto Service Mode Control

Hardware

4.11 DTMF SIGNAL SENDING

Description

Dual Tone Multi-Frequency (DTMF) signals are used with CO lines assigned for DTMF signaling. The duration of the DTMF signal can be adjusted from 40 to 990 milliseconds.

Operation

System

Operation of this feature is automatic when programmed.

Conditions

1. The system mutes the user's voice transmission to reduce interference while sending DTMF tones.

Programming

CO/IP
1. CO Line Signal Type (PGM 141-Btn 5)

SYSTEM
1. Inter Digit Timer (PGM 181-Btn 7)
2. DTMF Duration Timer (PGM 182-Btn 9)

Related Features

Dial Pulse Signaling

Hardware

4.12 IP ADDRESS DIALING

Description

If allowed, users may place calls using an IP path. The system accepts user dialed digits as the IP address for the called party. When dialing an IP call, the asterisk, '*', is used as the "dot" between bytes of the IP address.

Operation

iPECS Phone

To place an IP Call

1. Lift the handset or press the **[SPEAKER]** button.
2. Press **{IP GRP}** button or dial IP Group access code.
3. Dial 'xxx * xxx * xxx * xxx', use '*' as the dot in the IP address.
4. Press '#' to complete dialing.

SLT

To place an IP Call

1. Lift the handset.
2. Dial IP Group access code.
3. Dial 'xxx '*' xxx '*' xxx '*' xxx', use '*' as the dot in the IP address.
4. Press '#' to complete dialing.

Conditions

Programming

STATION

1. CO/IP Group Access (PGM 117)
2. Direct IP Call (PGM 122-Btn 1)

Related Features

Hardware

4.13 IP TRUNKING

4.13.1 H.323 v4 Service

Description

When assigned to support H.323 protocol, VoIP channels provide protocol conversion between H.323 v4 and the iPECS protocol or SIP. This permits the VoIP channel to connect to external H.323 networks or terminals and to support H.323v4 supplementary services. In addition, H.323 VoIP channels can register with an external H.323 GateKeeper to support Gatekeeper call routing.

Supplementary services are supported employing H.450.1 ~ H.450.12 standards, which define the following supplementary services:

H.450.1
H.450.2
H.450.3
H.450.4
H.450.5
H.450.6
H.450.7
H.450.8
H.450.9
H.450.10
H.450.11
H.450.12

Operation

System

Operation of H.323 Service is automatic

Conditions

Programming

STATION	1. Station IP Attributes (PGM 122)
BOARD	1. H323 VoIP Attributes (PGM 130)
CO/IP	1. CO/IP Line Group (PGM 141-Btn 1)
	2. CO VoIP Mode (PGM 141-Btn 11)

Related Features

System Networking
SIP Service
Remote Device Zone Management

Hardware

VOIM8 or VOIM24

4.13.2 SIP Service

Description

When assigned to support SIP (Session Initiation Protocol), VoIP channels provide protocol conversion between SIP and the iPECS protocol or H.323. This permits the VoIP channel to connect to external SIP networks for call services. In addition, to the IETF RFC-3261 Session Initiation protocol draft standard, iPECS VoIP channels support other SIP related RFCs including:

RFC-2617	HTTP Authentication, Basic & Digest
RFC-3515	Refer Method
RFC-3264	Offer/Answer Model
RFC-3265	SIP Basic Call Flow Examples
RFC-3891	SIP “Replaces” Header

Using the SIP database assignments, the system will register and authenticate with the SIP proxy server permitting the system to interoperate employing SIP to establish, manage and terminate real-time voice sessions with external parties.

Operation

System

Operation of SIP Service is automatic

Conditions

Programming

STATION
1. SIP User ID Table (PGM111-Btn 19)
2. Station IP Attributes (PGM 122)
3. Station SIP Attributes 2 (PGM 126, Web only)

BOARD
1. SIP Attributes (PGM 133, Web only)

CO/IP
1. CO/IP Line Group (PGM 141-Btn 1)
2. CO VoIP Mode (PGM 141-Btn 11)

Related Features

System Networking
H.323 v4 Service

Hardware

VOIM8 or VOIM24

4.14 IP WAN Dialing After Answer

Description
The iPECS system permits sending and receiving DTMF signals after connecting to an external VoIP party. The DTMF signal can be DTMF tone, Text String, or DTMF protocol (H.323 specification) based on the system programming.

Operation
System
Operation is automatic based on the system database.

Conditions
1. The connected VoIP party must transmit DTMF digits in the mode selected in the system database otherwise; the DTMF digits will not be recognized.

Programming

Related Features

Hardware

4.15 ISDN (Integrated Service Digital Network)

Description
The iPECS system supports both BRI (Basic Rate Interface) and PRI (Primary Rate Interface) ISDN circuits. Both North American standard 23B+1D channel and ETSI 30B+2D channel configurations are supported through the selection of the gateway Module, the T1/PRI supports NA standards while the PRIM supports the ETSI standards.

4.15.1 ISDN AOC (Advice of Charge) (N/A Australia)

Description
When ISDN Advice of Charge service is provided from the ISDN, the iPECS system will deliver charge information for display in the LCD of iPECS Phones and include the AOC in SMDR records. AOC is implemented in accordance with ETSI ISDN AOC Specifications.

Operation

System

ETSI standard AOC operation is automatic.

Conditions

1. AOC information, which is implemented based on ETSI AOC standard, can be sent during call set-up (AOC-S), during the call (AOC-D) or at the end of call (AOC-E).
2. This feature may not be available in the specific ISDN service area or may be a subscription service.

Programming

SYSTEM	1. SMDR attributes (PGM 177)
CO/IP	1. Advice of Charge (PGM 143-Btn 9)

Related Features

SMDR (Station Message Detail Recording)
Call Cost Display

Hardware

ISDN gateway Module

4.15.2 Calling/Called Party Identification

Description

The iPECS system receives calling party identification in the ISDN call Set-up message, CLIP-Calling Line Identification Presentation. The answering party identification, which may be different from the called party, is received in the ISDN connect message, COLP (Connected Line Identification Presentation). When provided, the LCD of iPECS Phones displays the identification, which is included in call records. The received identification may be sent to a selected serial port.

iPECS Phone Display:

```
LINE RINGING
CLI 03438502821
```

CLI Serial output:
record at ring start:
AA : BBBBBBBBCCC

record at ring stop
AA : BBBBBBBBCCC : DDDDDDDDD(EEE) -> FFF

record at answer
AA : BBBBBBBBCCC : DDDDDDDDD(EEE) -> FFF

Where:

AA – ISDN CO line number
BB…B – Line Identification received
CCC – Called station
DD…D – Speed Dial name
EEE – Speed Dial bin number
FFF – Answering station

The system will also compare the identification to the Speed Dial bins. If a match exists, the Name of the Speed Dial bin may be displayed in place of the number, CO/IP Name display.

The system will send calling and answering party identification in the appropriate messages to the ISDN based on the database. Identification messages may be restricted, not reported, to the far-end user. Calling Line Identification Restriction and Connected Line Identification Restriction may be enabled in the system database or by **{CLIR}** and **{COLR}** Flex buttons.

Operation

System

Operation of this feature is automatic.

iPECS Phone

*To program **{CLIR}** button,*

[PGM] + **{FLEX}** + **[PGM]** + ‘81’ + **[SAVE]**

*To program **{COLR}** button;*

[PGM] + **{FLEX}** + **[PGM]** + ‘82’ + **[SAVE]**

To activate CLIR or COLR, before placing or answering an ISDN call

1. Press the **{CLIR}** or **{COLR}** Flex button.

Conditions

1. This feature may not be available in the specific ISDN service area or may be a subscription service.

Programming

STATION

1. CLIP LCD Display (PGM 114-Btn 1)
2. COLP LCD Display (PGM 114-Btn 2)

CO/IP	1. COLP Table Index (PGM 143-Btn 1)
	2. Type of Calling Number (PGM 143-Btn 4)
ISDN/ICLID	1. CLIP Print to Serial Port (PGM 200-Btn 2)

Related Features

Hardware

ISDN gateway Module

4.15.3 Keypad Facility

Description

The ISDN Keypad Facility Information Element (IE) may enable the user to activate certain ISDN services (e.g. Off-Net Forward). To access this facility, the station must be enabled and have a Flex button for **{KEYPAD FACILITY}**. When activated, the digits dialed by a user are sent in the Keypad Facility IE instead of the Called Party Number IE.

Operation

iPECS Phone

To program **{KEYPAD FACILITY}** button;

[PGM] + **{FLEX}** + **[PGM]** + '89' + **[SAVE]**

To activate the keypad facility while on an ISDN line

1. Press the **{KEYPAD FACILITY}** button.
2. Dial desired digits, other actions disable Keypad facility feature.

Conditions

1. This feature can be activated or deactivated only after a CO line (ISDN) is seized.
2. Once activated, the system will continue to send dialed digits as Keypad Facility IE messages regardless of ISDN messages. Thus, in the connected mode, DTMF tones are not sent to the connected party, only the keypad message is sent.
3. If a Speed Dial bin includes **[MSG/CALLBK]** button as the first entry, the remaining digits in the Speed Dial number will be sent in a Keypad Facility IE.
4. This feature may not be available in the specific ISDN or may be a subscription service.
5. The system can handle only a single Call Reference; services that require handling of two simultaneous Call References, cannot be supported.

Programming

STATION	1. Flex Button Assignment (PGM 115)

Related Features

Station Speed Dial
System Speed Dial

Hardware

iPECS Phone
ISDN gateway Module

4.15.4 Multiple Subscriber Number (MSN)

Description

MSN enables up to 8 Subscriber Numbers to be allocated to one Basic Rate Interface allowing Direct-Dial-In call delivery. Sub-Address information is included as additional digits in the Called Party Number Information Element. These additional digits allow the system to ring a specific station in the MSN pool of phone numbers, iPECS-Micro, iPECS-50 and MFIM100: 250, MFIM300: 500, MFIM600: 1000 and MFIM1200:1500. When MSN or Sub-address information is received, the system references the MSN Table for an index to the Flexible DID Table, which determines call routing.

In addition, the MSN pool can be utilized for multiple calling number of station. If a station is assigned a loop button that references a specific MSN entry, the specific MSN will be served as a calling number of the station when make an outbound call over ISDN or SIP using the assigned button.

Operation

System

Inbound Call

Automatic by programming of Called Telephone Number and Index of Flexible DID Table in MSN pool.

Outbound Call

Press the assigned MSN Loop Button, and dial over ISDN or SIP.
The assigned MSN telephone number will be served for calling number. And it is recorded by account code of call in SMDR.

Programming MSN Table Entries (PGM 202)

Called Telephone Number : Calling Number Digits. Mandatory.
CO Line Range : Utilization is Optional. Only the specified CO lines are employed when make an outbound call using a MSN button if CO line range is specified. If CO line range is not specified then it employs CO lines as same method as normal loop button does.

Button Assignment

Using Station PGM button : **[PGM]**+**{FLEX}**+**[PGM]**+'85'+'#'+MSN digits+**[SAVE]**

(CF. MSN DIGITS SHOULD EXIST IN ONE OF MSN TABLE ENTRIES)

Using Station Flexible Button Programming :

1. by Keyset
   Dial '6'(MSN BUTTON) for Type option,
   input MSN digits and [Save]
2. by WebAdmin
   Type – LOOP
   Value - MSN digits

Print option: as an account Code of Call Record

PGM177–PRINT MSN: ON/OFF:

Conditions

1. If the Called Party Number from the ISDN does not match an MSN Table entry, the received digits are treated as defined for DID calls to determine an index to the Flexible DID Table.
2. The MSN Table employs the CO Line entry as a quick look-up reference. If the CO line number is programmed in the MSN Table, then only those entries with the CO line number are searched, otherwise the entire MSN Table is searched for a match to the Called Party Number.
3. When programs {MSN loop button} the 'MSN digits' should be matched to one entries of MSN pool. The button references an index of MSN entries.
4. MSN loop button is overwritten by a individual CO line button if it is exist. On the other hand, MSN loop button is not overwritten by CO Group button.

Programming

STATION	1. MSN Wait (PGM 114-Btn 12) 2. Flexible Button Assignment (PGM 115)
CO/IP Lines	1. CO/IP Attributes III (PGM 143)
ISDN/ICLID	1. ISDN Attributes – I (PGM 200) 2. MSN Table (PGM 202)

Related Features

Direct Inward Dial (DID)

Hardware

ISDN gateway Module

4.16 ISDN SUPPLEMENTARY SERVICES

In many cases, the ISDN service provider offers enhanced services, to which a user may subscribe. The iPECS system allows access to these ISDN "Supplementary Services" implemented under the ETSI regime as described below.

4.16.1 ISDN Call Deflection

Description

When the ISDN Supplementary Service "Call Deflection" is supported, a user can forward incoming calls on the ISDN line directly through the ISDN without the need to establish a connection through the system.

When Call Deflection is used, the system sends a call-deflection request message with the telephone number to receive the call to the ISDN. The ISDN then sends incoming calls to the desired telephone number. In this case, the system does not set-up a CO line to CO line (Unsupervised Conference) connection for the call.

Operation

iPECS Phone

To activate ISDN Call Deflection to an external number

1. Lift the handset or press the **[SPEAKER]** button to receive dial tone.
2. Press the **[FWD]** button.
3. Dial Forward condition ('1'~'4', '#')
4. Press **[SPEED]** button and desired bin number.
5. Replace the handset, return to idle.

To deactivate ISDN Call Deflection

1. Press flashing **[FWD]** button, Call Forward will deactivate and the **[FWD]** button LED is off.

Conditions

1. The ISDN must support the Call Deflection Supplementary Service as defined by the ETS300-202/206/207 standard protocol.
2. ISDN lines that support Call Deflection must be assigned in the system database, PGM 143, Btn 7.

Programming

CO/IP 1. ISDN-SS CD (PGM 143-Btn 7)

Related Features

ISDN Supplementary Services

Hardware

ISDN line
iPECS Phone

4.16.2 ISDN Malicious Call Id Request

Description

When the ISDN supports the Malicious Caller Id supplementary service, iPECS can request the CLID from the ISDN. The CLID is saved and output in the SMDR call record. Malicious Caller Id is activated during an ISDN call and requires a **{MCID REQUEST}** Flex button to access the feature.

Operation

iPECS Phone

*To program an **{MCID REQUEST}** button;*

[PGM] + **{FLEX}** + '*0' + **[SAVE]**

To request MCID, while on an incoming ISDN call

1. Press the **{MCID REQUEST}** button.

Conditions

1. The ISDN must support the Malicious Caller Id Supplementary Service as defined by the EN300-130 standard protocol.
2. If the MCID request is successful, the SMDR call record will include the CLID returned from the ISDN and the characters "MT". If the request fails, the characters "MF" are included in the record.
3. If the system is programmed to save SMDR records, the MCID is stored in the system memory as part of the call record.

Programming

CO/IP 1. MCID Request Code (PGM 109-Btn 1)

SYSTEM 1. SMDR Save (PGM 177-Btn 1)

Related Features

ISDN Supplementary Services

Hardware

ISDN line
iPECS Phone

4.17 Private Line

Description
One or more iPECS Phone users can be assigned exclusive use of a CO line or lines. These "Private" lines are assigned to CO line Group 00 by the iPECS system and an appearance **{CO}** button must be assigned at the user's iPECS Phone.

Operation
iPECS Phone

To place an outgoing Private line call
1. Press the **{CO}** button assigned as the Private line, dial tone is received from the CO line.
2. Dial the desired number.

To answer an incoming Private line call
1. Press the flashing **{CO}** button.

Conditions
1. Private lines will not recall to the Attendant station.
2. Private lines are not subject to Call Pick-Up except by an assigned Secretary with an appearance of the Private line.
3. When put on hold, a Private line will recall the user after the appropriate Hold timer. The user will receive recall for the Hold Recall time, and, if still unanswered, the user will continue to receive recall for the Attendant Recall time. If the call remains unanswered, at expiration of the Attendant Recall time, the call will be disconnected and the Private line returned to idle.
4. A Private line will follow Call Forwarding only if the receiving station has an appearance (**{CO}** button) for the Private line. However, calls on a Private line can be forward to the Voice Mail permitting a voice mail to be recorded.
5. IP channels cannot be assigned as Private lines.

Programming
STATION	1. Flex Button Assignment (PGM 115)
CO/IP	1. CO/IP Line Group (PGM 141-Btn 1)

Related Features
Hold
Call Forward
Executive/Secretary Forward
Call Forward, Preset

Hardware
iPECS Phone

4.18 MULTIPLE CLI

Description

This feature provides flexibility for CLI service.

In PGM 114, CLI Station Number(PGM 114 – F6) can be programmed.
Station CLI 2 ~ Station CLI 5 will be newly made.
By using these new CLIs, a specific CLI can be provided to the called party.
The name – [CLI Station Number] will be changed to [Station CLI 1].

In PGM 143 – Station CLI Type, CLI can be chosen between Station CLI 1 to Station CLI 5.
It will be newly made.
Station CLI 1 means Original CLI Station Number(PGM 114 – F6) is used.
Station CLI 2 ~ Station CLI 5 means newly added CLI Station Numbers are used.

In PGM 143 – CLIP table index, CLIP can be chosen.
The range is available from 00 – 50.
In case of CLIP table index 00 to 49, then normal CLI making way is applied.
In case of CLIP table index 50, then ONLY Station CLI Station Number(PGM 114 – F6) can be sent by CLI without adding any prefix.
The 50'th CLIP index will be used generally to all CLI cases(from Station CLI 1 to Station CLI 5).
The 50'th CLIP index name will be changed to [Station CLI].

Operation

Example - Station 110 has 5 CLI and related program is as followings.
1. CLI Station Number (Original CLI) - 110.
2. Station CLI 2 – 01234567.
3. Station CLI 3 – 01234567890123456.
4. Station CLI 4 – 110.
5. Station CLI 5 – 1234567890.
6. COLP Table index 01 – 450.
7. Area Code – 031.
8. In PGM 143, CO 1 is set to CLI 1, CO 2 is set to CLI 2 …. CO 5 is set to CLI 5.

Station 110 Call to outside via CO 1, CLIP table is 01.

CLI is 031 450 110.

Station 110 Call to outside via CO 2, CLIP table is 01.

CLI is 031 450 01234567.

Station 110 Call to outside via CO 2, CLIP table is 50(Station CLI).

CLI is 01234567.

Station 110 Call to outside via CO 3, CLIP table is 50(Station CLI).

CLI is 01234567890123456.

Net Slave station 110 seizes the transit out co-line 2. CLIP table is 01. PSTN CLI Method is set to ON.
CLI is 031 450 01234567.

Net Slave station 110 seizes the transit out co-line 2. CLIP table is 50. PSTN CLI Method is set to ON.
CLI is 01234567.

Conditions

1. New CLI Station Number length is Max 16.
2. When program new CLI in PGM 114, Character can be entered in each PGM for future req uirement. But, now it is recommended to enter only Numerals.
3. Default value for PGM 143 – Station CLI Type is CLI Index 1(Normal CLI).

Programming

STATION	1. PGM 114 – Station CLI 2 – Station CLI 5
CO/IP	1. PGM 143 – Choice CLI Type.(CLI 1 – CLI5)
	2. PGM 143 – COLP/CLIP Table.

Related Features

Hardware

5. IPECS PHONE

5.1 ANSWERING MACHINE EMULATION

Description

When a call is sent to a Voice mailbox, the associated station can be assigned to notify the user and allow the user to screen the call. Two methods of notification and call screening are provided, Ring or Speaker mode.

In the Ring mode, the user is notified by flashing of the AME (Answering Machine Emulation) Flex button. The user may press the Flex button to hear the caller as the voice message is stored. In the Speaker mode, when the call is sent to the Voice Mailbox, the caller's voice is automatically broadcast over the speaker of the user's iPECS Phone.

The user may terminate the screening leaving the caller in voice mail to record a message, talk with the caller and record the conversation in the mailbox, or answer the call and disconnect the Voice Mail.

The user's iPECS Phone must be assigned with an AME Flex button for proper operation.

Operation

iPECS Phone

*To assign an **{AME}** button:*

Ring Mode

[PGM] + **{FLEX}** + ‘564’ + ‘0’ + **[SAVE]**

Speaker Mode

[PGM] + **{FLEX}** + ‘564’ + ‘1’ + **[SAVE]**

To screen a call in the Ring mode;

1. Press the flashing **{AME}** button, the caller’s voice is broadcast over the station speaker and stored in the Voice Mailbox.

To stop the voice broadcast and leave the caller in Voice Mail;

1. Press the illuminated **[SPEAKER]** button.

To talk with the caller and record the conversation in Voice Mail;

1. Press the illuminated **[MUTE]** button.

To answer the call and cancel the voice message;

1. Press the illuminated **{AME}** button, the caller is connected and the Voice Mail disconnected.

Conditions

1. AME is supported only on an iPECS Phone and an **{AME}** Flex button must be assigned on the phone.
2. If the user answers the call using the **{AME}** button, the caller is connected in the normal manner, the Voice Mail is disconnected and, with the VMIM/VSF, any message recorded by the caller is not stored.

Programming

NUMBERING 1. AME Feature (PGM 109-Btn 2)

Related Features

External Auto Attendant/Voice Mail
VMIM/VSF Integrated Auto Attd/Voice Mail

Hardware

iPECS Phone

5.2 AUTO CALLED NUMBER REDIAL (ACNR)

Description
This feature allows a station user to request and have the system retry a busy or no answer external call until the call is connected or the feature is cancelled.
This feature is working on SLT also.

Operation
iPECS Phone

To activate ACNR while receiving busy, no answer

3. Press the **[REDIAL]** button.
4. Hang-up handset, or press **[SPEAKER]**.

To cancel ACNR while idle

1. Press flashing **[REDIAL]** button.

To cancel ACNR during an ACNR attempt

1. Lift the handset or press the **[MUTE]** or flashing **[REDIAL]** button.

SLT Phone

To activate ACNR while receiving busy, no answer

1. In SLT, outgoing call is established.
2. Outside destination is no answer or busy.
3. Press and release the hook-switch, the intercom dial tone should be heard
4. SLT user press ACNR code.
5. While on waiting state, SLT user hears MOH.

System

1. The system initiates the ACNR process, starting the ACNR Pause Timer.
2. At expiration of the timer, the system activates the station's speakerphone with the microphone in the mute mode.
3. The system attempts the previous call.
4. When the called party answers, the user may answer by lifting the handset or pressing the **[MUTE]** button to communicate with called party.

Conditions
1. Four timers and a retry counter can be programmed.

ACNR Pause Timer	Time allowed between ACNR retries.
ACNR Delay Timer	At expiration of Pause Timer, if no line is available, the system will wait for delay timer before retry.
ACNR Tone Detect	After dialing, the system will abandon retry if no tone or answer is detected within the Tone Detect time.
ACNR Retry Count	Count determines the number of times system will retry before ACNR is automatically cancelled.

2. The call will be placed on the same path as originally used. If the path is busy, an available CO/IP line in the same group will be seized.
3. The ACNR Retry Counter decrements by one when the system completes the dialed number.
4. When the ACNR Pause Timer expires, if the station is in a busy state, the ACNR Delay Timer is invoked.

5. Upon completion of dialing, the system will monitor the call for progress signals.
6. To preserve ACNR feature in activated state, SLT user does not go On-Hook.
7. While ACNR dialing or waiting state, if SLT user goes On-Hook, then SLT ACNR is deactivated, automatically.
8. During ACNR dialing SLT, cannot receive any new call.

Programming

SYSTEM
1. ACNR Delay Timer (PGM 180-Btn 8)
2. ACNR Pause Timer (PGM 180-Btn 9)
3. ACNR Retry Counter (PGM 180-Btn 10)
4. ACNR Tone Detect Timer (PGM 180-Btn 11)

NUMBERING PLAN
1. SLT ACNR Code (PGM 109-Btn 18 – 58#)

Related Features

LNR (Last Number Redial)
Speakerphone
Mute

Hardware

iPECS Phone

5.3 AUTO RELEASE OF [SPEAKER]

Description

After completion of certain features, the **[SPEAKER]** turns off automatically, returning the iPECS Phone to idle.

Operation

System

*Auto Release of **[SPEAKER]** operation is automatic for supported features, see conditions below.*

Conditions

1. This feature applies to all User and Attendant Programming except Custom Message, CO/IP line Disable and Version Display. Auto Release of **[SPEAKER]** also applies to features including Call Park, Call Back, Call Forward and CO/IP Queuing.
2. If, during Station User Programming, erroneous data is entered, error tone is received and the user must correct the error before the station will return to idle automatically.

Programming

Related Features

Hardware

iPECS Phone

5.4 Automatic Speaker Select

Description

iPECS Phones programmed for Auto Speaker Select can access a CO/IP line or an internal call by pressing the appropriate button without the need to lift the handset or press the **[SPEAKER]** button. Audio from the CO/IP line or called station is sent to the speaker as if the user pressed the **[SPEAKER]** button and the speakerphone's MIC is activate.

Operation

iPECS Phone

To access an internal or external system resource;

1. Press an assigned **{FLEX}** button.

Conditions

1. For iPECS Phones not equipped/assigned with speakerphone, the user must lift the handset to be heard.
2. Paging while on the speakerphone will cause feedback from the paging equipment. If Auto Speaker is enabled and a **{PAGE ZONE}** button is pressed, the display will show "LIFT THE HANDSET". To complete the page, the user must lift the handset within the predefined 5-second period or return to idle.

Programming

STATION 1. Auto Speaker Select (PGM 111-Btn 1)

Related Features

Hardware

iPECS Phone

5.5 Back Ground Music

Description

An iPECS Phone can receive audio, generally music, from an internal or external source while it is idle. Music from the source is received over the speaker and will be shut-off during ringing, pages, or while the station is off-hook.

Operation

iPECS Phone

To Receive Background Music,

1. Press **[PGM]** button
2. Dial '73' to BGM code
3. Dial 00-10 to select hear and select the BGM.
   00 – no BGM
   01 – Music 1
   02 – Music 2
   03 – VSF MOH 1
   04 – SLT MOH 1
   05 – SLT MOH 2
   06 – SLT MOH 3
   07 – SLT MOH 4
   08 – SLT MOH 5
   09 – VSF MOH 2
   10 – VSF MOH 3
4. Press **[SAVE]** to save your selection

Attendant

To Transmit BGM through an External Page Port from an Attendant

1. Press **[PGM]** button.
2. Dial '074' or '075', the Attendant Station Program code for External Page Port 1 or 2, respectively.
3. Dial 00-10 to select hear and select the BGM.
   00 – no BGM
   01 – Music 1
   02 – Music 2
   03 – VSF MOH 1
   04 – SLT MOH 1
   05 – SLT MOH 2
   06 – SLT MOH 3
   07 – SLT MOH 4
   08 – SLT MOH 5
   09 – VSF MOH 2
   10 – VSF MOH 3
4. Press **[SAVE]** to save your selection

Conditions

1. The speaker volume is adjustable at the station by using the **[VOL UP]/[VOL DOWN]** key of the iPECS Phone.
2. BGM is delayed 1 second after a return to idle state.
3. Silence is provided if no BGM source is assigned.
4. For remote devices, BGM must be locally provided; BGM is not sent from an MFIM to remote devices.

5. iPECS-Micro and iPECS-50 does not provide Music 2.
6. VSF MOHs must be recorded properly. VSF MOH2 and VSF MOH3 should be assigned prompt number in PGM 171 button 5 and 6.
7. SLT MOHs must be connected properly and should be assigned station number in PGM 171 button 4.

Programming

SYSTEM

1. BGM Type (PGM 171-Btn 1)
2. MUSIC Setting (PGM 171-Btn 3)
3. Assign SLT MOH Port (PGM 171-Btn 4)
4. VSF MOH2 (PGM 171-Btn 5)
5. VSF MOH3 (PGM 171-Btn 6)

Related Features

MOH (Music-On-Hold)
Internal/External & All Call Page

Hardware

BGM source properly connected to the MFIM, refer to the iPECS Description and Installation Manual, Section 4.4.2

5.6 CALL LOG DISPLAY

Description

Users of iPECS Phones with Soft keys can view a log of incoming, outgoing and missed calls on the display. A Flex button must be assigned as a **{CALL LOG}** button, which allows easy access to the Call Log menu.

Operation

iPECS Phone

*To assign a **{CALL LOG}** button:*

[PGM] + **{FLEX}** + **[PGM]** + '57' + **[SAVE]**

To access the Call Log menu;

1. Press the **{CALL LOG}** button.

STATION 101 (T)		
14 SUN 04		14:30
CALLED	DIALED	LOST

2. Using the Soft keys, select the desired call log (incoming, outgoing, or missed).

STATION 101 (T)
14 SUN 04 14:30
PREV NEXT SEND

3. Use Previous and Next soft keys to view the log contents.

Conditions

1. This feature applies to the LIP-7016D, 7024D, 7024LD, 8012D, 8024D and 8040L)

Related Features

Hardware

iPECS 7016D, 7024D, 7024LD, 8012D, 8024D, 8040L Phone

5.7 CO LINE NAME DISPLAY

Description

When a CO/IP call is received or a user seizes a CO/IP path, the CO/IP number is displayed on the LCD. If the CO/IP path is assigned a name and CO/IP Name Display is enabled, the CO/IP name is displayed instead of the CO/IP number.

Operation

iPECS Phone

To display the CO/IP line name while calling

1. Press the **{CO}**/**{IP}** button, or CO/IP Access code, LCD displays CO/IP line name.

Conditions

1. Each CO line and IP group is assigned a name of up to 12 characters in the system database.

Programming

CO/IP

1. CO Line Name Display (PGM 142-Btn 1)
2. CO Line Name Assign (PGM 142-Btn 2)

Related Features

Hardware

iPECS Phone w/Display

5.8 DID CALL WAIT

Description
If DID call is incoming to a station that is already call connected, this DID call is wait until the station is answering or the DID/DISA no answer timer is expired. To activate this feature, the ADMIN field of DID call wait must be set enable. If station has a flexible button of DID call wait, then it can set this feature via flexible button.

Operation
iPECS Phone

*To assign a **{DID CALL WAIT}** button:*

[PGM] + **{FLEX}** + **[PGM]** + '34' + **[SAVE]**

To activate/deactivate DID call wait from an iPECS Phone:

1. Press the **{DID CALL WAIT}** button.
2. Dial activate/deactivate code, '1' or '0' respectively.

Conditions
1. The DID call will follow the call routing defined in PGM code 167 after the expiration of the DID/DISA no answer timer expires.
2. The iPECS Phone must have an appearance button for the DID line.
3. Assigning the ICLID Timer, which enables ICLID routing, for a DID line, disables DID Call Wait.

Programming
STATION	1. DID Call Wait (PGM 113-Btn 9)
CO/IP LINE	1. ICLID Timer (PGM 142-Btn 14)
SYSTEM	1. DID/DISA Destination (PGM 167)
	2. DID/DISA No Answer Timer (PGM 181-Btn 2)

Related Features
Direct Inward System Access (DISA)
Direct Inward Dial (DID)
ICLID Call Routing

Hardware
iPECS Phone

5.9 DND – ONE TIME DND

Description
A station, which is ringing or receiving Off-hook muted ring, can press the **[DND]** button, to reject the call and terminate ringing. The station is placed in DND, ringing terminates and the call receives treatment based on the following precedence:

1. Previous or active Call Forward busy.
2. Preset Call Forward busy.
3. Station Call Coverage.
4. Direct Transfer to Voice Mailbox.
5. Return busy signal and disconnect.

When the station returns to the idle status, DND is cancelled and the **[DND]** LED is extinguished.

Operation

iPECS Phone

To activate One Time DND while ringing

1. Press the **[DND]** button, the **[DND]** LED lights, station goes to DND state.

System

Deactivation

1. When the station returns to idle, DND is disabled and the **[DND]** LED extinguishes.

Conditions

1. CO/IP recalls will override One Time DND.
2. The Attendant can override stations in One Time DND by using Camp-On or intrusion. The Attendant does not have One Time DND service.
3. One Time DND cancels existing Callback queues.

Programming

STATION	1. DND (PGM 111-Btn 3)
	2. Call Forward (PGM 111-Btn 2)
	3. Preset Call Forward (PGM 120)
	4. Direct xfer to Mail box (PGM 120-Btn 6)
SYSTEM	1. System Call Forward No Answer Timer (PGM 181-Btn 1)

Related Features

Call Waiting/Camp-On
DND (Do Not Disturb)
Call Forward
Call Forward, Preset
Station Call Coverage

Hardware

iPECS Phone

5.10 Flex Button Direct Speed Dial Assignment

Description

A user may program a telephone number directly to a Flex button without the need to assign the number to a Speed Dial bin. In this case, the telephone number is allocated to the highest numbered Station Speed Dial bin available.

Operation

iPECS Phone

To assign a telephone number to a Flex button;

1. Press the **[PGM]** button.
2. Press the desired Flex button.
3. Press the soft button below the "TEL NUM" display selection or, for iPECS 24D and 24DH phones, press the **[ICM]** button.
4. Press the CO/IP line button or dial the CO/IP line access code.
5. Dial the telephone number.
6. Press the **[HOLD/SAVE]** button
7. Dial the name to be associated with the number (optional).
8. Press the **[HOLD/SAVE]** button.

To place a call using the Flex button;

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the assigned Flex button.

Conditions

1. This feature is available to users of the iPECS 7016D, 7024D 7024LD, 8012D, 8024D or 8040L Phones only.
2. When a Flex button is assigned with a telephone number, the system will allocate the number to the highest available Station Speed Dial bin number. If no bin is available, the user will receive error tone when attempting to assign the telephone number.
3. The telephone number may include any of the special Speed Dial instructions **([MSG/CALLBK]**, **[FLASH]**, display security, etc.).

Programming

Related Features

Station Speed Dial

Hardware

LIP-7016D, 7024D, 7024LD, 8012D, 8024D and 8040L Phone

5.11 Flexible LED Flash Rates

Description

The flash rates used with the various Flex buttons on the iPECS phone can be adjusted on a system wide basis to meet the customer's needs. Up to 29 different functions can be assigned any one of 15 different flash rates.

Operation

System

System implements flash rates automatically based on database entries.

Conditions

1. Available Flash rates and functions, which can be assigned, are given in the iPECS Admin & Programming Manual.

Programming

SYSTEM 1. Button LED Flash Rates (PGM 170-Btn 1~29)

Related Features

Hardware

iPECS Phone

5.12 GROUP LISTENING

Description

All iPECS Phones have a built in speaker. If allowed, users may employ the speaker to monitor a call while using the handset to converse with the outside party. This enables a group of people in the room to listen to both parties in the conversation.

Operation

iPECS Phone

While on a call using the handset

1. Press the **[SPEAKER]** button, speaker activates, the speakerphone microphone will be muted while the handset is off-hook.

Conditions

1. While using the speakerphone, lifting the handset will turn off the speakerphone. To activate Group Listening, the **[SPEAKER]** button must be pressed while the handset is off-hook.
2. While in Group Listening Mode, pressing the **[MUTE]** button will cause the TX path from the handset to be muted. However, the distant end is still heard over both the handset receiver and the station speaker.

3. If full speakerphone operation is desired and available while in Group Listening Mode simply place the handset on-hook.

Programming

STATION 1. Group Listening (PGM 113-Btn 3)

Related Features

Speakerphone
Mute

Hardware

iPECS Phone

5.13 STATION INDIVIDUAL CALL ROUTING (ICR)

Description

Station ICR is an extension of Call Forward where the user establishes a routing scenario. Each of the ten scenarios defines rules to route incoming calls based on Time, Day of week, Date and Caller ID to a destination defined by the User. Each scenario is assigned a priority of 0 to 9. When an incoming call is received at the station, the System will search the ICR scenarios entered by the User, then the call will routed according to the destination in the highest priority matching scenario.

Operation

iPECS Phone

To create a scenario:

1. Press the **[PGM]** button.
2. Dial 24, the ICR menu or log on to the Station Web portal.
3. Select the desired Scenario number (0 - 9).
4. Select the type of Caller ID (0 – 5):
    - Type 0 – Station CID
    - Type 1 - All Station
    - Type 2 – CO CID
    - Type 3 - All col
    - Type 4 - All call
    - Type 5 - N/A
5. Enter the Scenario Start and End Date (YYMMDD).
6. Select day-of-week using Flex button:
    - Flex button 1 (MON) – Flex button 7 (SUN)
    - Flex 8 button (Holiday)

7. Enter scenario Start Time and End time (HHMM).
8. Select Type of destination (0 – 6):
   - Type 0 – Station
   - Type 1 - Hunt group
   - Type 2 - CO Line Number
   - Type 3 - CO Group number
   - Type 4 - CO Loop
   - Type 5 - CO Transit
   - Type 6 - N/A
9. Select Scenario Priority (0 to 9).

To activate ICR call forward:

1. Lift the handset or press the **[SPEAKER]** button to receive a dial tone.
2. Press the **[FWD]** button.
3. Dial the desired Call Forward code (0-4).
4. Dial 587, the Station ICR code.
5. Replace the handset, return to idle.

Conditions

1. Any station allowed access may define scenarios in the station's Web portal.
2. The conditions as listed under Call Forward may apply.

Programming

STATION

1. Call Forward (PGM 111-Btn 2)
2. Station Call Forward No Answer Timer (PGM 123-Btn 1)

SYSTEM

1. System Call Forward No Answer Timer (PGM 181-Btn 1)

Related features

Call Forward

Hardware

iPECS Phone

5.14 CALL PROFILE

Description

This function provides many flexibility of call re-routing.

If user is not in office, user is in another place, user can get the call by pre-programmed call profile table. Each station has its 3 call profile tables (table 1 ~ table 3), and each profile table can have up to 10 destinations.
In Station Program menu ([PGM] + [2] + [6] : CALL PROFILE), user can set or disable Call Profile Table. User can choose one Call Profile Table of three tables by this menu. If one Call Profile is selected, incoming call can routed by this Call Profile table.
In Station Program menu ([PGM] + [2] + [4] : ICR SCENARIO), user can configure Call Profile destinations scenario. Each destination scenario composed of Destination, Scenario Priority, Call profile table index(0:Deact 1-3:Call Profile table) and Call profile timer.

Operation

iPECS Phone

To create a Call Profile Scenario: (bold text step is necessary)

1. Press the **[PGM]** button.
2. Dial 24, the ICR menu or log on to the Station Web portal.
3. Select the desired Scenario number (0 - 9).
4. Select the type of Caller ID (0 – 5):
    - Type 0 – Station CID
    - Type 1 - All Station
    - Type 2 – CO CID
    - Type 3 - All col
    - Type 4 - All call
    - Type 5 - N/A
5. Enter the Scenario Start and End Date (YYMMDD).
6. Select day-of-week using Flex button:
    - Flex button 1 (MON) – Flex button 7 (SUN)
    - Flex 8 button (Holiday)
7. Enter scenario Start Time and End time (HHMM).
8. **Select Type of destination (0 – 6):**
    - Type 0 – Station
    - Type 1 - Hunt group
    - Type 2 - CO Line Number
    - Type 3 - CO Group number
    - Type 4 - CO Loop
    - Type 5 - CO Transit
    - Type 6 - N/A
9. **Select Scenario Priority (0 to 9) (0:highest priority).**
10. Select Forward From Net (0 / 1).
11. **Select Call Profile Table Index (0 - 3).**
12. **Select Call Profile Timer (10 – 60 sec).**

To activate ICR call forward:

1. Lift the handset or press the **[SPEAKER]** button to receive a dial tone.
2. Press the **[FWD]** button.
3. Dial the desired Call Forward code (0-4).

4. Dial 587, the Station ICR code.
5. Replace the handset, return to idle.

To change Call Profile in Local mode:

1. Press the [PGM] button.
2. Dial 26, the Call Profile menu or log on to the Station Web portal.
3. Select Call Profile Table(0-3). cf) (0:Deact 1-3:Call Profile table).

To change Call Profile in Remote mode:

1. Call to iPECS System (DISA CO Line).
2. Enter “#” to access voice mail box.
3. Enter Station Number.
4. Enter Auth code of station.
5. Enter “7” to set call forward
6. Enter “1” to set station call forward.
7. Enter “Station ICR Code”.
8. Enter “Call Profile Number” (0:ICR fwd, 1-3:Call Profile fwd).

Conditions

1. This feature is possible only Digital Co line.
2. This feature is applied only incoming call.
3. Ring Assigned feature does not applied with this feature.
4. This feature is applied not only Station Manual Call Forward but also Preset Call Forward (PGM120).
5. It is recommended hunt group is set last call profile destination.
6. Call Profile timer can be set from 10 sec to 60 sec.
7. First Call Profile timer is always 0 sec regardless of saved value. (A user can confirm this in station web admin – Station ICR – Overview)
8. Call Profile timer can be set from 10 sec to 60 sec.

Programming

STATION

1. Call Forward (PGM 111-Btn 2)
2. Station Call Forward No Answer Timer (PGM 123-Btn 1)

SYSTEM

1. System Call Forward No Answer Timer (PGM 181-Btn 1)

Related features

Call Forward
Station ICR

Hardware

iPECS Phone

5.15 INTERCOM SIGNALING MODE

Description

Each iPECS Phone can select the signaling mode used for incoming ICM calls while the station is idle. There are three signaling modes available.

H - Call announcing with Handsfree answerback:
When an ICM call is received, the user receives splash tone followed by the ICM caller's voice. The user may respond to the caller without the need to 'Lift the handset' or 'press the **[SPEAKER]** button'.

P - Call announcing with Privacy:
When an ICM call is received, the user receives splash tone followed by the ICM caller's voice. To respond the user must lift the handset or press the **[SPEAKER]** button.

T - Tone ringing:
An ICM call will cause the iPECS Phone to provide audible ICM ring tone. The user must lift the handset or press **[SPEAKER]** to answer. An SLT always functions in the Tone ring mode.

Operation

iPECS Phone

To change ICM Signaling Mode

1. Press **[PGM]** button, the **[SPEAKER]** button LED lights steady.
2. Dial Station User Program code '13', confirmation tone is received.
3. Dial the desired ICM Signaling Mode code ('1' for H, '2' for T or '3' for P).
4. Press the **[SAVE]** button.

Conditions

1. Message Wait, Callback, Call Forward and Attendant Override will ring in the tone mode, regardless of ICM Signaling Mode selected by the user.
2. The ICM signaling Mode Selection does not affect Page announcements.
3. The default ICM Signaling mode is Tone ring and the active mode is stored in battery-protected memory.

Programming

Related Features

Intercom Call (ICM Call)
Paging
Message Wait/Call Back
Call Forward
DND Override
Intrusion

Hardware

iPECS Phone

5.16 MUTE

Description

An iPECS Phone can turn off audio transmission from the handset, speakerphone or headset microphone, "Mic Mute".

Operation

iPECS Phone

To Mute the Microphone;

1. Press the **[MUTE]** button, the **[MUTE]** button LED is on and the microphone (Handset, Speakerphone, Headset) is muted; the connected party receives silence.

To activate the microphone;

1. Press the illuminated **[MUTE]** button, the **[MUTE]** button LED is off, and the microphone is activated, transmitting audio to the connected party.

Conditions

1. Changing from speakerphone to handset or vice versa during a mute condition will eliminate the mute status.
2. Returning to idle or placing another CO/IP or intercom call will change the mute status to its normal (active microphone) condition.

Programming

STATION

1. Headset Ring (PGM 111-Btn 8)
2. Speakerphone (PGM 111-Btn 9)

Related Features

Speakerphone
Group Listening
Headset Compatibility

Hardware

iPECS Phone

5.17 Off-Hook Signaling

Description
When a station, which is off-hook, receives a call or a CO/IP call rings into the system for the off-hook station, the station will receive the assigned Off-hook ring signal or, for ICM calls, a Camp-On, Voice-Over Announcement or Off-hook ring signal may be received.

The Off-hook ring Signal may be either a muted normal ring signal or a single tone burst based on the system assignment. This signal is delivered to the iPECS Phone speaker.

Operation
System

Operation of Off-hook ring signals is automatically controlled.

Conditions
1. While using the speakerphone, a Camp-On tone is provided over the speaker in place of the assigned Off-hook ring Signal.
2. Activating the DND or One-Time DND places the station in DND, terminating any Off-hook signaling.
3. Off-hook ring signals terminate when the call is answered, forwarded, or abandoned.
4. A station, which is receiving Off-hook ring signals, will receive normal ring upon return to idle status.

Programming
CO/IP	1. CO Line Ring Assignment (PGM 144)
System	1. Off-Hook Ring Signal Type (PGM 161-Btn 1)

Related Features
Call Waiting/Camp-On
CO/IP Ring Assignment
DND (Do Not Disturb)
DND - One Time DND

Hardware
iPECS Phone

5.18 ON-HOOK DIALING

Description

iPECS Phones equipped with a Speakerphone allow users to place as well as receive calls while the handset is on-hook. Once the user activates the speakerphone by pressing the **[SPEAKER]** button or Automatic Speaker Select, dial tone is received and the user may dial the desired number.

Operation

iPECS Phone

To activate On-Hook Dialing

1. Press the **[SPEAKER]** button, dial tone is received and the **[SPEAKER]** button LED lights.
2. Place desired call (dial station ICM number or select CO/IP path and dial).

Conditions

1. If the outgoing call is not answered, the user must press the illuminated **[SPEAKER]** button to return to idle.
2. When the speakerphone is used, the microphone is active unless the **[MUTE]** button is pressed and the **[MUTE]** button LED is On.

Programming

STATION

1. Auto Speaker Selection (PGM 111-Btn 1)

Related Features

Mute
Speakerphone
Automatic Speaker Select
Headset Compatibility
Group Listening

Hardware

iPECS Phone

5.19 Prime Line Immediately/Delayed

Description

When a user goes to an off-hook state, the system normally provides ICM dial tone. If desired, a station can be assigned to access a pre-assigned system resource (Prime Line). The Prime Line can be any of the Idle Line Settings:

- Seize a CO/IP Line,
- Call another station,
- Activate a Flex button feature.
- Call to pre-assigned Hunt group.
- Call to pre-assigned station Speed.
- Call to pre-assigned system Speed.

Prime Line access can be defined as immediate or delayed. When assigned immediate, upon an off-hook event, the system provides access to the Prime Line. With Delayed Prime Line, the station user receives normal Intercom dial tone for the Prime Line Delay timer and after the delay, the Prime Line is accessed.

Operation

iPECS Phone

To access the station's Prime Line

1. Lift the handset or press **[SPEAKER]** button and take no action, Prime Line as assigned will be accessed.

Conditions

1. Any of the station's Flex buttons may be assigned as the Prime Line. When the user lifts the handset or presses the **[SPEAKER]** button, the system will act as if the user had pre-selected the button prior to going off-hook.
2. Selection of another Flex button or Feature button just prior to an off-hook event will override the Prime Line assignment.
3. SLT can use the Flex Button Feature of Idle Line Selection with a Speed Bin Number.
4. When Delayed Prime Line is set, the user must wait, taking no action until the Prime Line is accessed. The user receives ICM dial tone during this period and may dial any valid numbering plan digit(s) or select a Flex button or feature button.
5. If the Prime Line Delay Timer is greater than Dial tone timer, the Delayed Prime Line will not activate. It will be necessary to reduce the delay timer or extend the Dial tone timer.

Programming

STATION	1. Prime Line setting (PGM 113-Btn 7)
	2. Idle Line Selection (PGM 121)
SYSTEM	1. Prime Line Delay Timer (PGM 182-Btn 6)

Related Features

Speakerphone

Intercom Call (ICM Call)
Station Flexible Buttons

Hardware

iPECS Phone

5.20 RING TONE DOWNLOAD

Description

The user can select one of 14 Ring tones so that the iPECS Phone ring can be distinguished from other nearby phones. Eight tones are stored in the iPECS Phone permanent memory. The first four tones are fixed and the 5th through 8th Ring tone can be downloaded from a library of 10 tones stored in the system's protected memory.

After downloading the tone from the system memory, it can be selected as the Differential Ring Tone.

Operation

iPECS Phone

To download a Ring Tone from the system memory:

1. Press the **[PGM]** button.
2. Dial 15, Ring Tone Download code.
3. Dial the desired Ring tone location, '5'~'8'.
4. Dial digits '00' through '10' to hear the Ring Tones.
5. Press the **[SAVE]** button.

To select the downloaded Ring Tone:

1. Press the **[PGM]** button.
2. Dial 11 for Intercom Ring tones, 12 for CO/IP Ring tones.
3. Dial the digit ('5'~'8') to select the Ring Tone
4. Press the **[SAVE]** button.

Conditions

1. The downloadable ring tone files are stored in system memory as *.wav files with a maximum length of 4 seconds. These files can be replaced as desired using the Web Upload function.

Programming

Related Features

Station User Programming & Codes
Differential Ring

Hardware

iPECS Phone

5.21 SAVE NUMBER REDIAL (SNR)

Description

The last dialed number on a CO/IP call may be stored (up to 48 digits) in a buffer for future redial. This number is saved in memory until the user requests a new number be stored. Numbers dialed for subsequent calls do not affect the Save Number buffer.

Operation

iPECS Phone

To save a dialed number, while on a CO/IP call

1. After dialing and before hanging up, press the **[SAVE]** button.

To save a dialed number, while on a CO/IP call using the LIP-8000 menu

1. After dialing and before hanging up, press the **[RIGHT NAVIGATION]** button.
2. Locate and press the **[SAVE]** soft button

To dial a saved number

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[SPEED]** button.
3. Dial #.

To save a dialed number, while on a CO/IP call using the LIP-8000 menu

1. Press the **[DIR]** soft button.
2. Press the **[SPEED]** soft button.
3. Dial '#'.

Conditions

1. The saved number can be a maximum of 48 digits.
2. Dialing the saved number will automatically seize the CO/IP line that was used for the original call. If the CO/IP line is busy, a CO/IP line from the same group will be selected and the saved number dialed. If all CO/IP lines from the group are busy, the user will receive All Lines busy tone and may queue
3. If user presses the **[SAVE]** button after seizing a CO/IP line without dialing, the Save Number Redial buffer will be erased.
4. If there is no **{CO}**/**{IP}** button, the call will be presented on a **{POOL}**, or **{LOOP}** button.
5. Save Number Redial is protected from power failure.
6. Manually dialing a Flash during a CO call will cause only those digits after the Flash to be stored and re-dialed as the Save Number Redial.

Programming

Related Features

Station Speed Dial
System Speed Dial

LNR (Last Number Redial)
CO Line Flash

Hardware

iPECS Phone

5.22 Silent Text Message

Description

Silent Text Messaging is used to respond to an OHVO call without disconnecting the existing call. Silent Text Messages are sent by pressing a pre-programmed message button or **[DND]** button.

Operation

iPECS Phone

*To program a **{TEXT MESSAGE}** button*

[PGM] + **{FLEX}** + **[PGM]** + '51' + Display Message code ('00'-'20') + **[SAVE]**

To respond to an Off-Hook Voice Over call with a Text message

1. Upon receiving announcement, press the desired **{TEXT MESSAGE}** or **[DND]** button, text message is sent to the display of the calling party, OHVO call is terminated.

Conditions

1. Silent Text Messaging employs the Custom Message/Pre-defined Display Messages.
   But, Display Message code in **{TEXT MESSAGE}** button is 01 ~ 09, the "STA IN DO NOT DISTURB" message is displayed.
2. When Silent Text Messaging is used to respond to an OHVO call, the existing call will not be disconnected.
3. When a SLT places the OHVO call, the receiving station will receive the "NO DISPLAY AVAILABLE" message and error tone to indicate Silent Text Messaging is not available to the SLT.

Programming

STATION 1. Voice Over (PGM 113-Btn 6)

Related Features

Call Waiting/Camp-On
Group Listening
Voice Over
Pre-defined & Custom Text Display Messages

Hardware

iPECS Phone

5.23 SPEAKERPHONE

Description

iPECS Phones equipped with speakerphone circuitry enable the telephone to be used hands-free in two-way conversations.

Operation

iPECS Phone

To activate the speakerphone

1. Press the **[SPEAKER]** button, **[SPEAKER]** LED lights steady.

To switch from Handset to Speakerphone

1. Press the **[SPEAKER]** button, **[SPEAKER]** LED lights steady.
2. Replace Handset, Speakerphone activated.

To terminate a speakerphone call

1. Press the **[SPEAKER]** button, **[SPEAKER]** LED extinguishes.

Conditions

1. If Automatic Speaker Select is enabled for the station, pressing a DSS, CO/LOOP/POOL/IP or Speed Dial button will automatically activate the speakerphone.
2. The **[MUTE]** button LED indicates the status of the Microphone, when lit the Microphone is inactive.
3. When Group Listen is enabled, pressing the **[SPEAKER]** button while using the handset will send audio to both the Handset and Speaker. However, only the Handset microphone is active. In order to activate the Speakerphone Microphone, the Handset must be placed on-hook.
4. Each iPECS Phone equipped with Speakerphone is allowed/denied Speakerphone operation in the system database.
5. When Headset operation is assigned for the station, the Speakerphone is disabled and the **[SPEAKER]** button activates the Headset audio path instead of the speaker.

Programming

STATION

1. Headset Ring (PGM 111-Btn 8)
2. Speakerphone (PGM 111-Btn 9)

Related Features

Mute
Group Listening

Automatic Speaker Select
Headset Compatibility

Hardware
iPECS Phone

5.24 STATION FLEXIBLE BUTTONS

Description
The iPECS Phone incorporates a field of “Flex” buttons as well as the fixed feature buttons. The Flex buttons are assigned in the system database to access features, functions and resources of the system. Specifically, Flex buttons can be assigned as:

- An Empty button has no system database assignment.
- A **{DSS/BLF}** button, used to place One-touch ICM call to a designated station and display status of the station.
- A Flex Numbering Plan, button activates the feature associated with the assigned digits from the Flexible Numbering Plan.
- A Speed Dial bin button accesses and dials the number from the assigned Speed Dial bin.
- A Pool button, accesses a CO/IP path from the CO/IP group designated for the Pool button. A Pool button is used primarily to provide an outgoing call appearance when a direct CO line appearance is not available. It may be used as the last choice for an incoming CO/IP call. The Pool button LED provides the status for the duration of the call.
- A Loop button provides an appearance for incoming CO/IP calls when a direct CO/IP appearance is not available. The Loop button LED provides the status for the duration of the call.
- A Station User Program Code button, access or activate the special features available with the Station User Program Codes, see section 5.24.
- A CO Line Appearance button provides access to the individual CO Line assigned to the Flex button. The CO Line button LED provides the status of the CO Line.

With the exception of CO Line buttons, Flex buttons can be assigned at the station by the end-user. If allowed in the database, the user can also assign or reassign CO Line buttons.

Operation
iPECS Phone
To assign a Flex button at the station
1. Press the **[PGM]** button.
2. Press the desired Flex button.
3. Dial the digits from the Flexible Numbering Plan.
4. Press the **[SAVE]** button.

Or,
1. Press the **[PGM]** button.
2. Press the desired Flex button.
3. Press the **[PGM]** button.
4. Dial the digits from the Station User Program Code or Fixed Number Plan.
5. Press the **[SAVE]** button.

Conditions

1. The **{LOOP}** and **{POOL}** buttons provide a status indication for the call as long as the station has supervisory control.
2. If there is no available **{CO}**/**{IP}** **{LOOP}** or **{POOL}** button for a call, the station will not ring for an incoming CO/IP call; transfers to the station immediately recall the initiator and recalls are routed directly to the Attendant.
3. A station may have multiple **{LOOP}** and **{POOL}** buttons.
4. When all CO Lines in a group are busy, any **{POOL}** buttons associate with the group will show busy, otherwise, **{POOL}** buttons will only display the status of an activate call.
5. **{POOL}** buttons will access a CO Line from the designated group using the Round Robin or Last Used method as assigned in the database.
6. The priority for the appearance of a CO call transfer is first a direct CO Line appearance (**{CO}**/**{IP}** button), if not available a **{LOOP}** appearance is employed, if not available a **{POOL}** appearance is used. If there is no appearance available, the transferring station recalls immediately.
7. Station User Program Codes are defined in section 5.24.

Programming

NUMBERING 1. Flexible Numbering Plan (PGM 105~109)

STATION 1. Flexible Button Assignment (PGM 115)

Related Features

Flexible Numbering Plan
Station User Programming & Codes

Hardware

iPECS Phone

5.25 STATION RELOCATION

Description

The iPECS Phone once registered can be re-located to any LAN port connected to the iPECS system without loss of any data or programming.

Operation

This feature is automatically activated.

Conditions

1. If an iPECS Phone requires replacement, the administrator must follow the terminal replacement procedure outlined in section 1 of the iPECS Series Admin & Maintenance Manual.

Programming

TABLES 1. Station IP Address (PGM 103-Btn 2)

Related Features

Hardware

iPECS Phone

5.26 STATION USER PROGRAMMING & CODES

Description

iPECS Phone users can program an array of functions and features, access status information and assign special features codes to Flex buttons. The Station User Program Codes used for these purposes are fixed as listed below.

Code	Description	Entries
10	Enblock Dial	
11x	Intercom Differential Ring	1~8
12x	CO/IP Differential Ring	1~8
13	ICM Signaling Mode	1: H, 2: T, 3: P
14x	Call Coverage Attribute	1~2
15x	Station Ring Download	5~ 8 + 00~19
19	Ear Mic Headset	6000 & 7000 phones only
21	Knock down Station COS	
22	Restore Station COS	Auth code
23	Walking COS	Auth code
24	ICR Scenario	Scenario Data
25	LIP Phone Statistics	Real-time Packet Monitoring
31	Station Msg Wait Retrieve	
32	CLI/IP Msg Wait Retrieve	

33	Register Authorization Code	Auth Code
34	DID Call Wait	
35	Exec/Sec MWI	1 On, 0 Off
36	Send SMS message	
37	Register Mobile Extension	tel number
38	Activate Mobile Extension	1 On, 0 Off
39	Register Mobile Ext CLI	tel number
41	Set Wake-Up Time	Once/Permanent & Hour/Min
42	Erase Wake-Up Time	
51	Custom/Pre-selected Msg	00~20
52	Register Custom Msg (00)	'Msg'
53	Create Conference Room	Room number (1~9)
54	Delete Conference Room	Room number (1~0)
57	**{Call Log Display}** button	
61	Headset/Speakerphone mode	0:H, 1:S
62	Select Headset Ring type	1:S, 2:H, 3:Both
*67	**{Call Coverage}** button	
71	LCD Display Mode	Domestic/English
72	Sys version display	
73	Select BGM source	(0~2)
74	User Name registration	'name'
75	Display Phone IP Address	
76	Change IP Address (Dot:*)	
77	Display MAC IP Address	
78	Change Mode	
79	Display Phone Version	
7*	Display Key numbers	
80	**{RECORD}** button	
81	ISDN **{CLIR}** button	
82	ISDN **{COLR}** button	
83	**{ACD DND}** button	
84	**{ACCOUNT CODE}** button	
85	**{LOOP}** button	
86	**{INTRUSION}** button	
87	**{ICM}** button	iPECS phones only
88	**{CAMP-ON}** button	
89	ISDN **{KEYPAD FACILITY}** button	
8#	**{OHVO}** button	
99	Push-To-Talk	
*0	Hot Desk Login	
**	Hot Desk Logout	
*7	Hunt Unconditional (forced) Call forward	
*8	Register Bluetooth device	
*9	Activate Bluetooth device	
90	**{Speed}** button	iPECS phones only

91	**{Conference}** button	iPECS phones only
92	**{Callback}** button	iPECS phones only
93	**{DND}** button	iPECS phones only
94	**{Flash}** button	iPECS phones only
95	**{Mute}** button	iPECS phones only
96	**{Monitor}** button	iPECS phones only
97	**{Redial}** button	iPECS phones only
98	**{Call Forward}** button	iPECS phones only

In addition, a Station User Program Menu display is provided by the iPECS Phone display to assist the user in programming the Station User Program Code features and functions. The **[VOL UP]**/**[VOL DOWN]** buttons are used to scroll through the menu items and the dial pad is used to enter a selection. The codes are also used to assign a function/feature to a Flex button. The Station User Program Menu display is graphically illustrated in the figure below.

USER PROGRAM MENU Displays:

First top-level Menu selection

[1] RING/NEWKEYSET
[2] COS

Under selection [1] Ring, select 1~5 as below

[1] STA RING TYPE
[2] CO RING TYPE

[3] ANSWER MODE
[4] CALL COVER ATTR

[5] STA RING DOWNLOAD
[1] STA RING TYPE

Under selection [2] COS, select 1~3 as below

[1] COS DOWN
[2] COS RESTORE

[3] WALKING COS
[4] ICR SCENARIO

[5] LIP KEYSET STAT
[1] COS DOWN

Next top-level Menu selection

[3] MSG RET/M-EXT
[4] WAKE UP TIME

Under selection [3] MSG RETRIEVE, select 1~8 as below

[1] MSG RETRIEVE METHOD
[2] MSG RETRIEVE EXAMPLE

[3] USER AUTH REGIST
[4] DID CALL WAIT

[5] CHOICE EXEC/SEC MSG
[6] SEND SMS MESSAGE

[7] REG MOBILE-EXT
[8] ACTIVE MOBILE-EXT

[9] REGISTER MOBIL CLI
[0] VM MOBILE NOTIFY

Under selection [4] WAKE UP TIME select 1~2 as below

[1] SET WAKE UP TIME
[2] WAKE UP DISABLE

Next top-level Menu selection

[5] MESSAGE
[6] HEADSET

Under selection [5] MESSAGE, select 1~2 as below

[1] SET PRE CUST MSG
[2] PGM CUSTOM MSG

[3] ACTIVE CONF-ROOM
[4] DEACTIVE CONF-ROOM

[5] MONITOR CONF_GROUP
[1] SET PRE:CUST MSG

Under selection [6] HEADSET select 1~2 as below

[1] HEADSET OR SPK MODE
[2] HEADSET RING MODE

Next top-level Menu selection

[7] SUPPLEMENTARY
[*] SYSTEM

Under selection [7] SUPPLEMENTARY, select 0~9 as below

[1] LCD DISPLAY LANGUAGE
[2] MFIM/E VERSION

[3] BGM
[4] REGISTER STA NAME

[5] DISP PHONE IP ADDR

[6] CHANGE PHONE IP ADDR

[7] DISP MAC ADDR
[8] CHANGE MODE

[9] STA MSG WAIT RETR
[*] DISP ADD-ON PACKAGE

Under selection [*] SYSTEM, access Admin

[#] ENTER ADMIN
[7] FORCED FWD TO DEST

[8] REGISTER BLUETOOTH
[9] BLUETOOTH USAGE

[0] HOTDESK LOGIN
[*] HOTDESK LOGOUT

Operation

iPECS Phone

To assign a Station User Program Code to a Flex button

1. Press the **[PGM]** button, the Station User Program Menu is displayed.
2. Press the desired Flex button.
3. Dial the desired Station User Program Code and additional inputs that may be required.
4. Press the **[SAVE]** button.

To activate a Station User Program Code feature or function

1. Press the **[PGM]** button, the Station User Program Menu is displayed.
2. If desired, use the **[VOL UP]/[VOL DOWN]** to display the desired menu item. or dial the desired Station User Program Code and additional inputs as required

Programming

Conditions

Related Features

Station Flexible Buttons
Temporary Station COS/Lock
Walking COS
Station Message Wait/Call Back
CLI Message Wait
Wake-Up Alarm
Pre-defined & Custom Text Display Messages
Headset Compatibility

Attendant Station Program Codes

Hardware

iPECS Phone w/Display

5.27 Two-Way Record

Description

An iPECS Phone user can record any active conversation to the station user's mailbox or to hard disk drive of an iPECS Phontage or UCS Client. All calls including incoming, outgoing, internal, external, conference, conference rooms and conference group calls can be recorded. A **{RECORD}** button must be assigned to access this feature.

Operation

iPECS Phone

*To assign a flexible button as a **{RECORD}** button*

[PGM] + **{FLEX}** + **[PGM]** + 80 + recording destination (optional) + **[SAVE]**

To activate Two-Way Record while on a CO/IP call

1. Press the **{RECORD}** button, record warning tone heard, and recording starts.

To stop Two-Way Record while on an CO/IP call

1. Press the **{RECORD}** again.

Or,

1. Hang-up, return to idle.

To manage the recordings, use the procedures outlined in the Phontage or UCS Client User Guide.

Conditions

1. The **{RECORD}** button LED will flash at 240 ipm while recording.
2. This feature is available when using the VMIM/VSF VM, the Feature Server or using an external AA/VM, which employs the SMDI communications mode. When an external AA/VM system employs the in-band (DTMF) mode, Two-Way Record is not available.
3. With the Feature Server or VMIM, internal calls can be recorded as well as external calls.
4. Use of this feature when the Two-Way Recording Warn Tone is disabled may be interpreted as a violation of federal, state, or local laws, and an invasion of privacy. Check applicable laws in your area before recording calls using this feature.
5. If a destination for the recording is not defined for the **{RECORD}** button, the Call Record Destination defined in PGM 112 is employed.

Programming

STATION

1. Two-way Record Privilege: ON (PGM 112-Btn 11)
3. Call Record Destination (PGM 112-Btn 21)
2. Auto Call Record (PGM 112-Btn 20)

SYSTEM

1. Two-way Record Warning Tone (PGM 161-Btn 19)
2. (Optional) without MCIM set 3-Way Conference Preference to LOCAL. (PGM 160-161 item 53)

Related Features

External Auto Attendant/Voice Mail
VMIM/VSF Integrated Auto Attd/Voice Mail
Conference
Conference Group
Multi-Party Voice Conference
Conference Room
Auto Call Recording

Hardware

iPECS Phone
VSF, VMIM, Feature Server or External SMDI based AA/VM system
Phontage or UCS Client

5.28 VOICE OVER

Description

This feature allows users of iPECS Phones, off-hook on a call (CO, IP or Intercom), to receive a voice announcement through the handset receiver with the existing call. The Voice Over is muted so as not to interfere with the existing conversation. The called station user may then respond to the calling party using Camp-On response or may use Silent Text Messaging to respond.

Operation

iPECS Phone

Placing a Voice Over (OHVO) while receiving busy

1. Dial '#' or press a pre-programmed **{OHVO}** button.
2. After splash tone, begin announcement.

Responding to a Voice Over announcement

1. Use Camp-On response procedure or Silent Text Messaging or One-Time DND.

SLT

Placing a Voice Over (OHVO) while receiving busy

1. Dial '#'.
2. After splash tone, begin announcement.

Responding to a Voice Over announcement

1. Use Camp-On response procedure.

Conditions

1. When the called station responds via Camp-On, all conditions and options available to Camp-On apply.
2. OHVO may be used to notify the called party of a transferred call (CO/IP Line or Intercom) by announcing the call then releasing to complete the transfer.
3. When a call is transferred via OHVO the receiving station will receive muted ringing after the transfer is complete.
4. The Pre-defined or Custom Text Display Message feature may be used in response to a Voice-Over announcement and the conditions associated with these features apply.
5. Both the originating and receiving station must be programmed to allow OHVO calls.
6. When Silent Text Messaging is used to respond to an OHVO call, the existing call will not be disconnected.
7. If the receiving station is in conference or using the Speakerphone or Group Listen, Voice Over is not available. Camp-On will be activated and a Camp-On tone sent to the receiving station.

Programming

STATION 1. Voice Over (PGM 113-Btn 6)

Related Features

Call Waiting/Camp-On
Group Listening
Silent Text Message
Pre-defined & Custom Text Display Messages

Hardware

iPECS Phone to receive Voice Over

5.29 SERIAL DSS

Description

LIP-8000 keyset supports Serial DSS to be connected through serial cable.

There are 3 types of serial DSS ; 12 button Serial DSS(LIP-8012DSS), 12 button Serial DSS with LCD(LIP-8012LSS), 48 button Serial DSS(LIP-8048DSS).

Operation

Connecting Serial DSS to LIP-8000 keyset

1. Check the station have the authority to use serial DSS in Admin.
2. Check the serial DSS power is off (only for LIP-8048DSS).
3. Power off the LIP-8000 keyset.
4. Connect serial DSS to keyset through serial cable.
5. Power on the DSS (only for LIP-8048DSS).
6. Power on the keyset.
7. Program DSS button on station program by pressing [PGM] + Flex button.

Or, program DSS button from web 'Flexible button PGM' menu.

Editing LSS Labe

1. After connecting serial LSS, enter ADM 129 and enter serial dss order and desired button number.
2. Edit Label string according to flex button feature.
3. If Label is saved and Flex button is programmed, the label string is displayed in the flex button field.

Conditions

1. Total number of Serial DSS per a system is limited. Refer the table below.

System	Max. number of Serial DSS	Max. number of LSS
iPECS-Micro	50	40
iPECS-50	50	40
iPECS-100	50	40
iPECS-300	100	60
iPECS-600	200	60
iPECS-1200	200	60

Above limitation includes Hot-desk Agent stations.

2. LIP-8004D doesn't support serial DSS.
3. Maximum 4 serial DSS boxes can be connected to one keyset.
4. LIP-8048DSS is powered by external adaptor, but others cannot use external adaptor. They should be powered from keyset or LIP-8048 DSS.
5. LIP-8000 keyset can supply power to maximum two DSS. For this, the keyset should use external adaptor. And LIP-8048 DSS also can supply power to maximum two DSS.
6. Linked pair Slave station cannot use the serial DSS.
7. If LSS label is not programmed, MFIM gives the default name of LSS feature.

8. If ICM number is programmed to LSS and the label is not programmed, destination station name is displayed. If the station doesn't have its name, STAXXX is displayed by default.
9. If Speed number is programmed to LSS and the label is not programmed, the Speed name is displayed. If the Speed bin doesn't have its name, SPDXXX is displayed by default.
10. When program DSS button feature of Label, 1st DSS starts from button 49, 2nd DSS starts from button 97, 3rd DSS starts from button 145, 4th DSS starts from button 193.
11. When using keyset admin serial DSS buttons don't work.
12. To program DSS button using keyset admin, press volume up/down button to scroll flexible button number index.
13. When DSS configuration is changed, the initial value of 48Btn DSS is as same as initial value of DSS map1 ~ map4, 12BTN DSS and 12BTN LSS doesn't have initial values.
14. If DSS configuration is changed and the phone restarts, programmed DSS data is initialized from the changed DSS.
15. To Control LSS contrast, press volume up/ down button in idle state. DSS contrast is changed with keyset LCD.
16. Once after serial DSS is connected to LIP-8000, the configuration DB is saved until different type of serial DSS is connected to the station.
17. Each station can have serial DSS usage authority option. If disabled, serial DSS configuration of the station is initialized and DSS is not installed to that station.

Programming

STATION

1. Serial DSS Usage (PGM 111 –Btn21)
2. Flex Button Assign (PGM 115)
3. Serial LSS Label Edit (PGM 129)

Related Features

Hardware

6. ATTENDANTS

6.1 ACTIVE CPU (MFIM) DISPLAY

Description

When the system is equipped for CPU Redundancy, the System Attendant can determine and change the active and standby CPU.

Operation

System Attendant

To view the active MFIM

1. Press the **[PGM]** button.
2. Dial '078' the Attendant Station Program code, the active CPU is displayed.

To Change Standby MFIM to Active MFIM

1. Press the **[PGM]** button.
2. Dial '078' the Attendant Station Program code.
3. Dial '#' to change standby MFIM to active MFIM.

Conditions

1. The system must be equipped with and properly programmed (**PGM** 161) for Redundant Processor operation for the Attendant to display or change the active MFIM.
2. The System Attendant can assign a Flex button to indicate when the back-up MFIM is active. When the Flex button is assigned as (**PGM** 078) the LED of the flex button will turn on indicating the back-up MFIM is active.

Programming

SYSTEM 1. CPU Redundancy Use (PGM 161-Btn 20)

Related Features

Redundant System Processor

Hardware

6.2 ALTERNATE ATTENDANT

Description

This feature allows an Alternate answer point while the Attendant station is in an unavailable mode. When in the unavailable mode, the next available Attendant in the Attendant group will receive Attendant calls and recalls.

Operation

Attendant

*To assign a flexible button to activate **{ALT ATD}** button*

[PGM] + **{FLEX}** + '562' + **[SAVE]**

To toggle Attendant Unavailable feature

1. Dial '562', the Alternate Attendant code or press **{ALT ATD}**.

Conditions

1. Alternate Attendant activates the DND feature at the Attendant station and affects all calls to the Attendant station.
2. A Flex button can be assigned to activate Alternate Attendant. The **{ALT ATTENDANT}** button LED indicates the status of the Alternate Attendant feature, On: Attendant unavailable.
3. A station, which is receiving calls forwarded from the System Attendant, cannot use the Alternate Attendant feature.
4. All except for one attendant can activate Alternate Attendant. When the last Attendant attempts to activate this feature, error tone is received.
5. An Attendant forwarded to an unavailable Attendant is also considered to be in the unavailable Attendant mode.
6. When there is a queued Attendant call, unavailable Attendant stations **[HOLD]** button will flash but no audible ring is provided and the station cannot retrieve the call. When an Attendant changes from unavailable to available status, any queued Attendant calls will be available to the Attendant.

Programming

Related Features

Hardware

6.3 Attendant Positions

Description

The iPECS -50 and 100 can have a maximum of four (4) Attendants. Other iPECS systems have capacity for five (5) Attendants. Each Attendant position must be equipped with an iPECS multi-button Phone and may include multiple DSS Consoles.

There are 2 different types of Attendant as follows,

- System Attendant - The first station in Attendant Group. In addition to Main Attendant features, the System Attendant is commonly used for System Attendant Programming, Admin Programming and Day/Night/Timed Mode control.
- Main Attendant – Attendant group stations other than the first station of Attendant Group.

Operation

Condition

1. As a default, the first Attendant (System Attendant) is assigned as Station 100 (logical number), and others are not assigned.
2. Attendant calls (dialing '9') and recalls are routed to first available attendant.

Programming

SYSTEM

1. System Attendant Assignment (PGM 164-Btn 1)
2. Main Attendants Assignment (PGM 164-Btn 2~4/5)

Related Features

Hardware

iPECS Phone

6.4 Attendant Recall

Description

Unanswered or abandoned CO/IP calls that remain unanswered for the Hold or Transfer Hold timer, as appropriate, will recall the station placing the call on hold. If the call remains unanswered for the assigned Recall time, the first available Attendant will also receive recall. The Attendant and station will receive the recall signal for the Attendant Recall Timer period after which the system will disconnect and return the CO/IP line to idle.

Operation

System

Attendant recall operation is automatic.

Conditions

Programming

SYSTEM

1. Attendant Recall Timer (PGM 180-Btn 1)
2. Hold Recall Timer (PGM 180-Btn 5)

Related Features

Hold
Call Transfer

Hardware

6.5 Attendant Station Program Codes

Description

Using the Attendant Station Program Codes, the Attendant can print SMDR and Traffic reports on-demand, assign Authorization Codes, control certain user features, record VMIM/VSF announcements, enable/disable Auto Service Mode Control, etc. Items are available using the

Program Code directly or scrolling the multi-level display menu. The following indicates the menu displays, including the digit for selecting the item, the item description and further required entries. The various levels of the display menu are indicated by indentation. For additional information, reference Appendix C of the iPECS Admin & Programming Manual. Note also, some Program Codes are only available to the System Attendant or stations allowed access to the Attendant Program code '0'.

[1] PRINT
    [1] SMDR
        [1] PRINT SMDR STA BASE
            station range input
        [2] DELETE STATION BASE
            station range input
        [3] DISPLY CALL CHARGE
        [4] ABORT PRINTING
        [5] PRINT LOST CALL
        [6] DELETE LOST CALL
    [2] TRAFFIC - System attendant only
        [1] PRINT ALL SUMMARY
            enter Analysis time & type
        [2] PRINT ALL PERIDICLLY
            enter Analysis time, type & Print time
        [3] ABORT PERIDIC PRINTING
        [4] PRINT ATD TRAFFIC
            enter Analysis time & type
        [5] PRINT CALL SUMMARY
        [6] PRINT CALL HOUR

        [7] PRINT H/W USAGE
            enter Analysis time & type
        [8] PRINT CO SUMMARY
            enter Analysis time & type
        [9] PRINT CO HOUR
            enter CO group number
[2] COS
    [1] SET ICM ONLY MODE
        enter station range
    [2] RESTORE COS
        enter station range
[3] AUTHORIZATION
        System Attendant only
    [1] REGISTER AUTHORIZATION
        enter station
    [2] ERASE AUTHORIZATION
        enter station
[4] DATE AND TIME

[1] LCD DATE TIME SET
System Attendant only
[2] LCD DATE MODE CHAGE
System Attendant only
[3] LCD TIME MODE CHAGE
System Attendant only
[4] ATD SET WAKE UP TIME
enter station range
[5] ATD WAKE UP DISABLE
enter station range
[6] Set Time from ISDN Message
enter 1= on, 0= off
[7] ATD HOTDESK LOGIN
[8] ATD HOTDESK LOGOUT
[5] MESSAGE
[1] SET PRE:CUSTOM MSG
enter station range & Msg (00~20)
[2] DND/FWD/MSG CANCEL
enter station range
[3] CUSTOM DISP MSG REG
enter Msg (11~20)
[4] Monitor Conference Room
[5] Delete Conference Room
[6] REC VSF ANNCEMENT
enter VMIM/VSF sequence number (001-999)
[7] SUPPLEMENTARY
[1] REG STATION NAME
enter station number
[2] ISOLATE CO FAULT
[3] AUTO D/N/T PGM
select enable/disable
[4] EXT PORT#1 BGM EN/DI
select 0-2
[5] EXT PORT#2 BGM EN/DI
select 0-2
[6] LCD Display Language
select language
[7] PTT Login, station range
enter station range

Operation

Attendant

To activate an Attendant Station Program Code feature or function

1. Press the **[PGM]** button, the Attendant Station Program Menu is displayed.

2. Dial '0' to access the Attendant Station Program codes.
3. Enter desired code or use the **[VOL UP]/[VOL DOWN]** button to display the desired menu item and enter the desired code.
4. Dial additional inputs, if required.

Condition

Programming

SYSTEM

1. System Attendant (PGM 164-Btn 1)
2. Main Attendants Assignment (PGM 164-Btn 2~4/5)

Related Features

SMDR (Station Message Detail Recording)
Traffic Analysis
Temporary Station COS/Lock
Authorization Codes (Password)
System Clock Set
Pre-defined & Custom Text Display Messages
VMIM/VSF Integrated Auto Attd/Voice Mail
Dial-by-Name
Auto Service Mode Control
Back Ground Music
Station User Programming & Codes

Hardware

iPECS Phone

6.6 CABINET ALARM

Description

The Cabinet Alarm alerts the Attendant of a fault in the fans or power supplies in a cabinet. The alarm notification displays the cabinet number. The cause of the fault, cabinet fan, PSU or PSU fan is shown on the Cabinet Attributes Web Admin page (**PGM** 197). The alert may be terminated at the Attendant's phone by dialing the Alarm Stop code or, if assigned, pressing the **{ALARM STOP}** button.

Operation

When an alarm condition occurs, the Attendant phone will display:

CABINET XXX STS FAULT

System Attendant

*To assign a Flex button as an **{ALARM STOP}** button to terminate the Alarm Signal:*

[PGM] + **{FLEX}** + '565' + **[SAVE]**

To terminate an Alarm Signal while idle:

1. Dial the Alarm Stop code, 565, confirmation tone is received and the display returns to normal.

Or,

1. Press the **{ALARM STOP}** button.

Conditions

1. The PSU serial port must be connected to the gateway Module or MFIM identified in the Cabinet Attributes Web Admin page (PGM 197).
2. If the PSU is connected to a WTIM, refer to the Description and Installation Manual for proper DIP-switch settings; DIP-switch 3 must be set to the Off position.
3. The notification identifies the cause of the alarm as FAN1 or FAN2 for the Main cabinet fans, PSU1 or PSU2 for a power supply fault or PSU1 FAN or PSU2 FAN.

Programming

SYSTEM 1. Cabinet Attribute (PGM 197)

Related features

Alarm Signal/Door Bell
Emergency Call Attendant Alert

Hardware

6.7 CALL FORWARD, ATTENDANT

Description

As with other stations, Attendants can forward calls to other stations in the system. Calls may be forwarded unconditionally, on busy or no answer.

Operation

Attendant

To activate Call Forward

1. Lift the handset or press the **[SPEAKER]** button to receive intercom dial tone.
2. Press the **[FWD]** button.
3. Dial code ‘0’ ~ ‘5’, the Call Forward type.
4. Dial the station number to receive forwarded calls.

To deactivate Call Forward

1. While idle, press the **[FWD]** button.

Or,

1. While Off-hook, press the **[FWD]** button.
2. Dial #.

Condition

1. The Conditions of the Call Forward feature apply.
2. If the System Attendant activates Unconditional Call Forward, the receiving station will receive Attendant calls and recall ring. In addition, if the receiving station is an iPECS Phone, the user will be able to activate features normally reserved for a Main Attendant.
3. The system requires at least one Attendant be available. The last available Attendant cannot activate Call Forward to an Attendant that has activated the Alternate Attendant feature.

Programming

Related Features

Call Forward

Hardware

6.8 CALL FORWARD CO/IP OFF-NET

Description

The System Attendant can forward incoming CO/IP calls to a remote “Off-Net” location. Calls are forward via a Speed Dial bin. When a call is received, the system will automatically place a call using the Speed Dial number, dialing the number and connecting the incoming call in an Unsupervised conference.

Operation

System Attendant

To activate CO/IP Off-Net Call Forward

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[FWD]** button.
3. Dial '5', the Off-Net Forward feature code.
4. Dial CO/IP access code to forward, the incoming call.
5. Dial the Speed Dial bin to be used to place the outgoing call, the LED of any Off Net forwarded **{CO}**/**{IP}** button at the Attendant stations will flutter at a rate of 240 ipm.

To deactivate CO/IP Off-Net Call Forward

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[FWD]** button.
3. Dial '5', Off-Net Forward code.
4. Dial CO/IP access Code.
5. Dial '#'.

Conditions

1. The System Attendant can forward any CO/IP line, CO/IP Group or all CO/IP calls using the appropriate CO/IP dial access code. In addition, a Flex button assigned to a CO/IP line or group may be forwarded using the Flex button.
2. The conditions of the Unsupervised conference feature apply.
3. The conditions of the Call Forward Off-Net feature apply.
4. When enabled, calls forwarded Off net will receive the Off net forward prompt.
5. When enabled, calls forwarded Off net will receive the DTMF repeat tone.
6. To utilize off-net call forward of incoming analog CO line to other analog CO line, a valid Open Loop Detect Timer should be assigned by programming for the analog CO lines to prevent CO line lock up when analog CO lines are in use.

Programming

STATION

1. Call Forward (PGM 111-Btn 2)
2 Off-Net Forward (PGM 111-Btn 14)

SYSTEM

1. Off-Net Prompt Usage (PGM 160-Btn 11)
2 DTMF Repeat Tone Usage (PGM 160-Btn 15)
3. System Attendant Assignment (PGM 164-Btn 1)
4. Unsupervised Conference Timer (PGM 182-Btn 5)
5. Open Loop Detect Timer (PGM 142-Btn 13)

Related Features

Call Forward
Unsupervised Conference

Hardware

6.9 CALL/QUEUING

Description

Any station can call the Attendant by dialing the Attendant Call code '9'. When an Attendant call experiences a busy, the call is queued to the Attendant group. The call will be delivered to the first available Attendant.

Operation

To call the Attendant

1. Dial '9', the Attendant Call Code.

Condition

1. Call routing order follows the order of entry in the Attendant Assignments program.
2. The Attendant is informed of a queued Attendant call by the **[HOLD]** button LED flashing. The calling intercom party will receive ring-back tone or MOH, as specified.
3. Calls to the Attendant's station intercom number are sent to the Attendant station dialed as with any intercom call.
4. When an attendant calls another busy attendant by dialing station number, busy tone is received and camp-on is available.

Programming

SYSTEM

1. Attendant Assignment (PGM 164)
2. MOH (PGM 171-Btn 2)
3. Attendant Call Queuing Tone (PGM 160-Btn 1)

Related Features

Attendant Positions
Intercom Call (ICM Call)

Hardware

6.10 DAY/NIGHT/TIMED/SCENARIO RING MODE

Description

The system clock automatically controls the Ring Mode. Ring assignments are applied based on the time of day and day of week. Four modes of ring (Ring Assignments) are provided, Day, Night ,Timed and Scenario.

The Attendant controls the system Ring Service mode changing from Auto Service Mode to Day, Night ,Timed or Scenario service mode. Based on the service mode selected, different ring assignments, COS and answering privileges are invoked for the system users.

Operation

System Attendant

To change Day/Timed/Night Ring Mode manually;

1. Press the **[DND]** button.
2. Dial 1~5. (1: Day mode, 2: Night mode, 3: Timed mode, 4: Auto Service mode, 5: Scenario mode)
3. If you set Scenario mode(5), you should set scenario group(MFIM100/300/600 : 01 ~16, MFIM1200: ~32).
4. Press the **[HOLD/SAVE]** button.

To set Day/Timed/Night Ring/Scenario Mode automatically (Auto Service Mode Control)

1. Press **[PGM]** button, **[PGM]** button LED flashes 60 ipm, **[SPEAKER]** LED lights steady.
2. Dial the Attendant Station Program code '#8' to toggle between manual and Auto Mode control.

Conditions

1. Only Attendants can change Day/Timed/Night/Scenario Ring Mode for the system manually and program the Auto Ring Mode Selection Table.
2. Stations receive incoming ring for CO lines based on the database assignments and the system mode (Day/Night/Timed/Scenario) when the call arrives.
3. When Auto Ring Selection Table is programmed, the ring, COS and CO/IP access mode are changed automatically based on the times assigned in the table.
4. The System Attendant always has manual control of the system mode by enabling/disabling the Auto Service Mode Control.
5. If you want to set manually Scenario mode in attendant, you should program Scenario group of System call routing table(PGM 251).

Programming

CO/IP
1. CO Line Ring Assignment (PGM 144)

SYSTEM
1. External Control Contact (PGM 168)

TABLES
1. Auto Ring Mode Selection Table (PGM 233)
2. System Call Routing (PGM 251)

Related Features

System Clock Set
CO/IP Ring Assignment

LBC (Loud Bell Control)
Dialing Restrictions
Auto Service Mode Control
System Call Routing

Hardware

6.11 DSS CONSOLES

Description

The system allows an unlimited number of DSS/LS Consoles to be installed in the system. Up to 1 DSS/DLS consoles may be associated with a station connected to an iPECS-Micro, and up to 3 DSS/DLS consoles may be associated with a station connected to an iPECS-50 or 100 and up to 9 may be associated with a station for all other system configurations. Each button on the console can be assigned as CO/IP line, DSS, or feature button.

For ease of programming, each console is assigned to a button 'map' in the system database. The user or Administrator may then change individual Flex buttons as desired. Operation of the DSS/BLF Console Flex buttons is the same as Flex buttons on the iPECS Phone.

Operation

Conditions

1. Each DSS/DLS Console is assigned to operate in connection with a specific station.
2. There is no limit to the number of DSS/DLS units in a system beyond the basic system capacities.
3. Assignments for the DSS/DLS button maps are detailed in the iPECS Admin & Programming manual. Note that the individual button assignments can be changed as with the iPECS Phone Flex buttons.

DSS/DLS button maps

iPECS-Micro

	MAP 1
Flex 1	Atd Override
Flex 2	All Call Page
Flex 3	Call Park 01
Flex 4-	Station Group 1
Flex 5	Camp-On
Flex 6	Int All Call Page
Flex 7	Call Park 02
Flex 8	Station Group 2
Flex 9	Release
Flex 10	Ext All Call Page

Flex 11	Call Park 03
Flex 12	Station Group 3
Flex 13 ~ 38	Station 100 – 125

iPECS-50 & 100 Maps 1 to 3

MAP 1	
Flex 1	Atd Override
Flex 2	All Call Page
Flex 3	Call Park 01
Flex 4-	Station Group 1
Flex 5	Camp-On
Flex 6	Int All Call Page
Flex 7	Call Park 02
Flex 8	Station Group 2
Flex 9	Release
Flex 10	Ext All Call Page
Flex 11	Call Park 03
Flex 12	Station Group 3
Flex 13 ~ 48	Station 100 – 135

MAP 2	
	iPECS-100
Flex 1 ~ 34	Station 136~169
	iPECS-50
Flex 1 ~ 14	Station 136~149
MAP 3	
Flex 1 ~ 42	CO 01~42
	iPECS-Micro
Flex 1 ~ 5	CO 01~05

DSS/DLS button maps

iPECS-300 & 600 & 1200 Maps 1 to 9

MAP 1	
Flex 1	Intrusion
Flex 2-	All Call Page
Flex 3 –	Call Park 01
Flex 4-	Station Group 1
Flex 5 –	Camp-On
Flex 6-	Int All Call Page
Flex 7 –	Call Park 02
Flex 8-	Station Group 2
Flex 9 –	Release

Flex 10-	Ext All Call Page
Flex 11 –	Call Park 03
Flex 12-	Station Group 3
Flex 13 ~ 48	Station 100 – 135

MAP 2	
Flex 1 ~ 48	Station 136~183

MAP 3	
Flex 1 ~ 48	Station 184~231

MAP 4	
Flex 1 ~ 48	Station 232~279

MAP 5	
Flex 1 ~ 48	Station 280~327

MAP 6	
Flex 1 ~ 48	CO 001~048

MAP 7	
Flex 1 ~ 48	CO 049~096

MAP 8	
Flex 1 ~ 48	CO 097~144

MAP 9	
Flex 1 ~ 48	CO 145~192

Programming

STATION

1. Station Type (PGM 110-Btn 1)
2. DSS/DLS Map (PGM 110-Btn 2)
3. Flexible Button Assignment (PGM 115)

Related Features

Attendant Positions
Station Flexible Buttons

Hardware

6.12 Disable Outgoing CO/IP Access

Description
The System Attendant can place CO/IP lines out-of service, disabling outgoing calls on the CO/IP path. This is normally done should an undetected fault interrupt service on a CO/IP path. Incoming calls continue to be processed normally.

Operation
System Attendant

To disable/enable Outgoing CO/IP access (toggle)

1. Press the **[PGM]** button.
2. Dial '072', the Attendant Station Program code.
3. Press the **{CO}** button of the line(s) to be disabled, confirmation tone is heard and the status for the selected line(s) is changed.

Conditions
1. If the desired CO/IP line is in use, the System Attendant may still disable the CO/IP line. The feature will take effect after the desired CO/IP line goes to idle.
2. Once the line is disabled, all Attendant appearances for the disabled CO/IP line will flutter at 240 ipm, other stations will indicate the CO/IP line as busy, LED is On.
3. The CO/IP line outgoing access status is stored in battery-protected memory in case of a power failure.
4. Multiple CO/IP lines may be enabled/disabled without redialing the Attendant Station Program code. Confirmation tone is heard after each CO/IP line is enabled/disabled.
5. When the system detects a fault on an analog CO line, the CO line is disabled for outgoing access automatically.
6. Incoming calls on a disabled CO/IP line will continue to operate normally.

Programming
SYSTEM 1. System Attendant (PGM 164-Btn 1)

Related Features
Attendant Positions

Hardware

6.13 DND OVERRIDE

Description
A station in the DND mode generally cannot receive an incoming call. The Attendant and the Secretary station of an Executive/Secretary pair however may override the DND status to signal the station of an awaiting call.

Operation
Attendant
To activate DND Override while receiving DND tone
1. Press the **{ATD INTRUSION}** button, the call signals at DND station.

Conditions
1. An Attendant may use Override to transfer a CO/IP call to a station in DND.

Programming

SYSTEM	1. Attendant Assignment (PGM 164)

Related Features
Intrusion
DND (Do Not Disturb)
Executive/Secretary Forward

Hardware

6.14 EMERGENCY CALL ATTENDANT ALERT

Description
When a station places a call to an Emergency number, the attendant receives an emergency call alert. The alert includes an alert tone and display of the emergency call information, which

continues until the Attendant resets the alert. The information includes the calling station number, time and date. The system stores the most recent emergency calls (up to 16). The Attendant can review the history at any time. To control atctivation or de-activation Emergency call Attenant alert feature, configure the option at PGM 161.

Operation

System

Operation of the alert is automatic.

Attendant

*To assign a Flex button as an {**ALARM STOP**} button to terminate the alert:*

[PGM] + **{FLEX}** + '565' + **[SAVE]**

To terminate an alert signal while idle:

1. Dial the Alarm reset Code, 565, confirmation tone is received and the Alert terminates.
   Or,
   Press the **{ALARM STOP}** button.

To view the Emergency Call History at the System Attendant phone:

1. Lift the handset or press the **[SPEAKER]** button.
2. Press the **[PGM]** button.
3. Dial '08', the Emergency Log code. The display shows the first emergency call logged in the history.

EMERGENCY STA NO XXXX	
MM/DD HH:MM	(xx)

4. Press the **[VOL UP]** and **[VOL DOWN]** buttons to scroll through the call history.
5. If you review a history then the item will be mared as below.

EMERGENCY STA NO XXXX	
MM/DD HH:MM	READ (xx)

6. If you review all history then the alarm will not provided when the station go to idle.

Programming

STATION	1. Emergency CO/Group (PGM 112, BTN 18)
TABLES	1 Emergency Code Table (PGM 226)
SYSTEM	1 EMR CALL ATD NOTIFY (PGM 161, BTN 24-6)

Conditions

Related Features

Alarm Signal/Door Bell
Emergency Call E-911 (caller location) Support
Emergency Call

Hardware

6.15 EZ-ATTENDANT

Description

The ez-Attendant is a Windows based PC application that provides a visualization of the Attendant functionality to simplify Attendant control of features and functions including displays of call, user and system status. ez-Attendant operates in conjunction with the Attendant's iPECS Phone to simplify operation and expand features and functions available to the Attendant.

For further information on ez-Attendant, refer to the ez-Attendant Installation and User Guide.

Operation

System Attendant

Operation of the ez-Attendant is provided in the ez-Attendant Installation and User Guide.

Conditions

1. iPECS ez-Attendant requires installation of a system Lock-key.

Programming

SYSTEM 1. Attendant Assignment (PGM 164)

Related Features

Hardware

6.16 FEATURE CANCEL

Description

The System Attendant can cancel features such as DND, Call Forwarding and Pre-defined or Custom Messages that are active at other stations.

Operation

System Attendant

To deactivate DND/Call Forward/Pre-selected Message for other stations

1. Press the **[PGM]** button.
2. Dial '052', Attendant Station Program code.
3. Dial the desired station range or the same station number twice for a single station.
4. Press the **[SAVE]** button, confirmation tone is heard and Attendant station returns to idle status.

Conditions

Programming

SYSTEM 1. System Attendant (PGM 164-Btn 1)

Related Features

Call Forward
DND (Do Not Disturb)
Attendant Positions
Pre-defined & Custom Text Display Messages

Hardware

6.17 INTRUSION

Description

An Attendant can intrude upon an active station conversation. When the Attendant intrudes an Intrusion Tone is provided, if assigned, and a conference is established between the Attendant, station, and the CO/IP party.

Intrusion can only be activated using an **{ATD INTRUSION}** button.

Operation

Attendant

*To assign an **{ATD INTRUSION}** button*

[PGM] + **{FLEX}** + **[PGM]** + '86' + **[SAVE]**

To activate attendant intrusion while receiving busy on an Intercom call

1. Press **{ATD INTRUSION}** button, Intrusion Warning Tone is provided to the busy station.

Conditions

1. If an Attendant or Secretary of an Executive/Secretary pair presses the **{ATD INTRUSION}** button while receiving DND tone, the system will activate DND Override.
2. The conditions of the Conference feature apply.

Programming

STATION 1. Override Privilege (PGM 113-Btn 4)

SYSTEM 1. Privacy (PGM 161-Btn 3)
2. Privacy Warning Tone (PGM 161-Btn 4)

Related Features

Hardware

6.18 LCD DISPLAY FORMAT CONTROL

Description

The System Attendant can select the format of the time and date provided to the LCD of all iPECS Phones in the system.

The System Attendant can select (toggle between) two formats for both time and date. The formats are:

Date: Month/day/year or Year/month/date
Time: 12 hour or 24 hour (military)

Operation

System Attendant

To Change LCD Date Format (toggle)

1. Press the **[PGM]** button.
2. Dial '042', the Date Display Format program code.

To Change LCD Time Format (toggle)

1. Press the **[PGM]** button.
2. Dial '043', the Time Display Format program code.

Conditions

Programming

SYSTEM
1. System Attendant (PGM 164-Btn 1)
2. LCD Date Display Mode (PGM 169-Btn 1)
3. LCD Time Display Mode (PGM 169-Btn 2)

Related Features

Attendant Positions

Hardware

6.19 SYSTEM CLOCK SET

Description

The System Attendant can set the system Time/Date.

Operation

System Attendant

To set the system clock

1. Press the **[PGM]** button.
2. Dial '041', the Attendant Station Program code.
3. Dial six (6) digits for the Date (MM/DD/YY) or **[SAVE]** to skip the Date.
4. Dial four (4) digits for the Time (HH/MM) or **[SAVE]** to skip the Time setup.
5. Press the **[SAVE]** button, confirmation tone is heard and Attendant station returns to idle status.

To set the system clock through ISDN message

1. Press the **[PGM]** button.
2. Dial '046', the Attendant Station Program code.
3. Dial '1' to set DATE/TIME using ISDN message.
4. Press the **[SAVE]** button, confirmation tone is heard and Attendant station returns to idle status.

Conditions

1. The time is entered as a 24-hour (military) clock, 24-hour mode.
2. If an NTP server is assigned, the system will check the time every ten (10) minutes. If the system time is more than 10 seconds off of the NTP time, the system time is reset. This will override the Attendant setting.

Programming

SYSTEM

1. Network Time/Date (PGM 161-Btn 12)
2. System Attendant (PGM 164-Btn 1)
3. System Time (PGM 178-Btn 1)
4. System Date (PGM 178-Btn 2)
5. DST Enable (PGM 178 – Btn 3)
6. DST Start & End Time (Web only)
7. NTP Active (PGM 195 – Btn 1)
8. NTP Sever address (Web only)
9. Std system time, local Time Zone (Web only)

Related Features

LCR (Least Cost Routing)

SMDR (Station Message Detail Recording)
Auto Service Mode Control
Automatic System Time Synchronization
Automatic System Daylight Savings Time
Day/Night/Timed/Scenario Ring Mode

Hardware

7. SLT

7.1 BROKER CALL

Description

Broker Call allows an SLT user to engage in two (2) calls, alternating between the two parties, so that the conversation with each party is private.

There are two types of Broker Call, Transfer and Camped On.
Transfer Broker Call: 2nd Call is originated by SLT user.
Camped On Broker Call: 2nd Call is delivered to the SLT through a Camp-On.

Operation

SLT

To activate a Transfer Broker Call

1. Make or receive an intercom or external call.
2. Momentarily press the hook-switch, intercom dial tone received and active call is placed in Exclusive hold state.
3. Place second call.
4. To alternate between calls momentarily press the hook-switch.

To activate a Camp-On Broker Call

1. Make or receive an intercom or external call.
2. Receive a Call Waiting/Camp-On tone.
3. Momentarily press the hook-Switch, intercom dial tone received and the active call is placed on Exclusive Hold.
4. Dial the Camp-On Answer feature code '600', camped on call is connected

To alternate between the calls

1. Momentarily press the hook-switch.
2. Dial the Camp-On Answer feature code '600'.

Conditions

1. After a hook-switch flash, if the call results in an error, busy, no answer or an abnormal state, the SLT user may momentarily press hook-switch to retrieve the held call.

2. During a Transfer Broker Call, if the SLT user goes on-hook, the Broker Call parties are connected completing a Call Transfer.
3. During a Transfer Broker Call, if the active caller disconnects from the SLT user, the held party, if another station, is connected to the SLT. If the held party is an CO/IP call, the SLT user receives error tone and may go on-hook to receive recall and retrieve the held call
4. During a Camp-On Broker Call, if the SLT user goes on-hook, the active call is disconnected and the held call recalls to the SLT.
5. During a Camp-On Broker Call, if the active party disconnects from the SLT, the SLT user receives error tone. The SLT user may momentarily press the hook-switch to retrieve the held party or go on-hook and receive recall.
6. If the SLT user presses the hook-switch twice in less than 2 seconds, a 3-way conference is established.
7. If after a hook-switch "Flash", the user takes no action for the dial tone timer, the SLT will receive error tone. If the SLT goes to an on-hook state, the SLT will receive recall ring automatically.

Programming

Related Features

Message Wait/Call Back
Call Waiting/Camp-On
Exclusive Hold
Call Transfer
Conference

Hardware

7.2 HOWLER TONE

Description

When an SLT station goes off-hook and does not initiate dialing in the Dial tone timer duration, delays dialing between digits in excess of the inter-digit time or stays off-hook at the completion of activating a feature or program, the station will receive howler tone as an error indication and the call attempt will be abandoned. In order to complete the call, the user must return to the on-hook state and restart the call.

Operation

System

The system will deliver howler tone automatically, as required

Conditions

1. Howler Tone is sent after a period, of about 30 seconds of error tone.
2. Lock-out occurs when howler tone starts.

Programming

STATION 1. Howling Tone To Stn (PGM 111-Btn 5)

Related Features

Intercom Lock-Out

Hardware

7.3 SLT Message Wait Indication

Description

All SLT devices will receive a “Stutter” dial tone as an audible Message Wait Indication. In addition, industry standard Message Waiting telephones may be connected to the system. Software will cause the lamp to flash when a messaging is waiting.

Operation

System

The system switches the 90 VDC lamp On and Off for assigned SLTs indicating a Message Wait.

Conditions

1. The system switches a 90 VDC supply On and Off to flash the SLT’s neon lamp.
2. Although the SLT Battery Feed is removed during the 90 VDC On cycle, the system will recognize an SLT Off-hook event.
3. The SLT must incorporate a 90 VDC neon lamp that is connected directly across the tip and ring of the voice network.

Programming

STATION 1. Station Type (PGM 110-Btn 1)

Related Features

Message Wait/Call Back

Hardware

SLT w/90 VDC Neon lamp

7.4 SLT NAME ENTRY

Description

A SLT user has the capability to program the user name so that a calling user with an LCD can see the name instead of the station number.

Operation

SLT

To register the name at the SLT

1. Lift the handset.
2. Dial ‘561’, the SLT Programming code, confirmation tone is heard.
3. Dial ‘74’, the SLT Name Program Code.
4. Enter name, refer to Station Speed Dial, Alphanumeric Chart.
5. Momentarily depress the hook-switch, receive confirmation tone.

To delete the name at the SLT;

1. Lift the handset.
2. Dial ‘561’, the SLT Programming code, confirmation tone is heard.
3. Dial ‘74’, the SLT Name Program Code.
4. Momentarily depress the hook-switch, receive confirmation tone.

Conditions

Programming

Related Features

Dial-by-Name
Station Speed Dial

Hardware

7.5 Transfer CLI to SLT

Description
A SLT phone can be received CLI from internal caller instead of station number by programming when it's tranfering.

Operation

SLT

To transfer to SLT when it's CO call

1. An User answers a call from CO with CLI
2. The User is transfering to another SLT
3. The SLT can see CLI of CO instead of station number

Conditions
1. If this function works well, the system should be set ORI in ADM 114-B18.

Programming
Station 1. Transfer CLI to SLT (PGM 114-Btn 18)

Related Features

Hardware

7.6 SLT Flash Mode

Description
SLT Flash works as following option.
0 : Flash Transfer - Flash detected, then the line is held and the line goes to waiting state.
1 : Flash Drop - Flash detected and Line is disconnected.
2 : Flash Ignore - Flash detected, but Ignored.
3 : Hold Release - Flash detected, then the line is held and the line goes to waiting state.
And the SLT user goes on-hook, then the held line is disconnected, not recalling.

Operation

Conditions

Programming

PGM 113 – SLT Flash Mode

Related Features

Hardware

8. SIP EXTENSION

8.1 REGISTRATION

Description

iPECS-LIK supports standard protocol equiped SIP Phone including series of LG-Ericsson SIP Phone Extension.

Operation

SIP Phone Self Programming

Network Configuration

1. IP mode : Static(Fixed) / DHCP
2. Subnet Mask
3. Default gateway IP address
4. IP address
5. DNS IP address
6. Prifiling (for Wireless)

SIP Server Configuration

1. Proxy IP address : MFIM IP address
2. Proxy IP port : 5060
3. Domain : MFIM IP address
4. Registration : ON
5. Registration Timer : 30 ~ 3600 second (more than 10 minute recommended)
6. Local UDP/TCP/TLS port : 5060 or other value
7. Signaling/Transport Mode : UDP (or TCP or TLS)

Line(User) Configuration

1. SIP Account :
   - Display Name (Optional) : Station Name (this will be applied to MFIM – Station Name).
   - User Name (Mandatory) : Station Number (this should be same as MFIM – Device Login / Station User Login (443) / ‘Desired Number’)
   - Authorization Name (Mandatory) : Login ID (this should be same as MFIM – Device Login / Station User Login (443) / ‘ID’)
   - Authorization Password (Optional) : Login Password Login ID (this should be same as MFIM – Device Login / Station User Login (443) / ‘Password’)

Call Preferences

1. Call Wait : ON / OFF (When on BUSY, accept other call setup or not)
2. Call Forward
3. DTMF Type (Mandatory) : one of INFO type (After registration to system, SIP Data / SIP Phone Attributes(211) / ‘DTMF Type’ – set the same type as SIP Phone)
   c.f) only support INFO type
4. CODEC
5. Call Blocking … and so on

Other Configurations

1. Answer Mode
2. Key, Tone
3. Digit Map (if SIP Phone support dial by Digit Map solution) :
   If you can utilize SIP Phone's programmable Dial Digit Map, you can make immediate sending out dials w/o press [SEND] or '#' key.
   <Refer to below>

- Digit Map (for Call Setup in SIP Phone)
  : you can program Digit Map according to System Numbering Plan
  Example) in case of 'LIP8002/88xx' SIP Phone type
  x.T|0|xx#|xxx#|1xxx|2xxx|3xxx|xxxx+#
  Explanation)
  '9' : dial '9' for attendant call, you can specify other single digit if you want it to be sent immediately when press the digit without ending by '#'
  '1xxx', '2xxx', '3xxx' : extension numbering, you can specify other 2 or 3 or 4 or more digits starting with a specific prefix digit or digits
  'xx#', 'xxx#', 'xxxx#' : dial the other 2, 3 or 4 digits and press '#' to send
  'xxxx + #' : dial the other more than 4 digits and press '#' to send
- Digit Map (for 2nd Dial Tone) : Play self dial tone by SIP Phone
  Example) in case of 'LIP8002/88xx' SIP Phone type
  0 // '0' is CO access code in the first CO group

System(MFIM) Programming

1st Registration

1. A SIP Phone will be registered as an Extension of LIK system
2. Check Extension Number : System ID & Numbering Plans / Flexible Station Number (105). You can use an Extension Number that is not assigned by other Extension or an empty Number. As example below, you can use '1050' because it is not yet assigned by other Extension or '1051' because it is unused and empty number.

Station Number	IP Address	MAC Address	New Station Number
1050			1050

Just check if extension number is available. Or you can change or arrange numbers to get a new extension number you wish to use by new registeration.

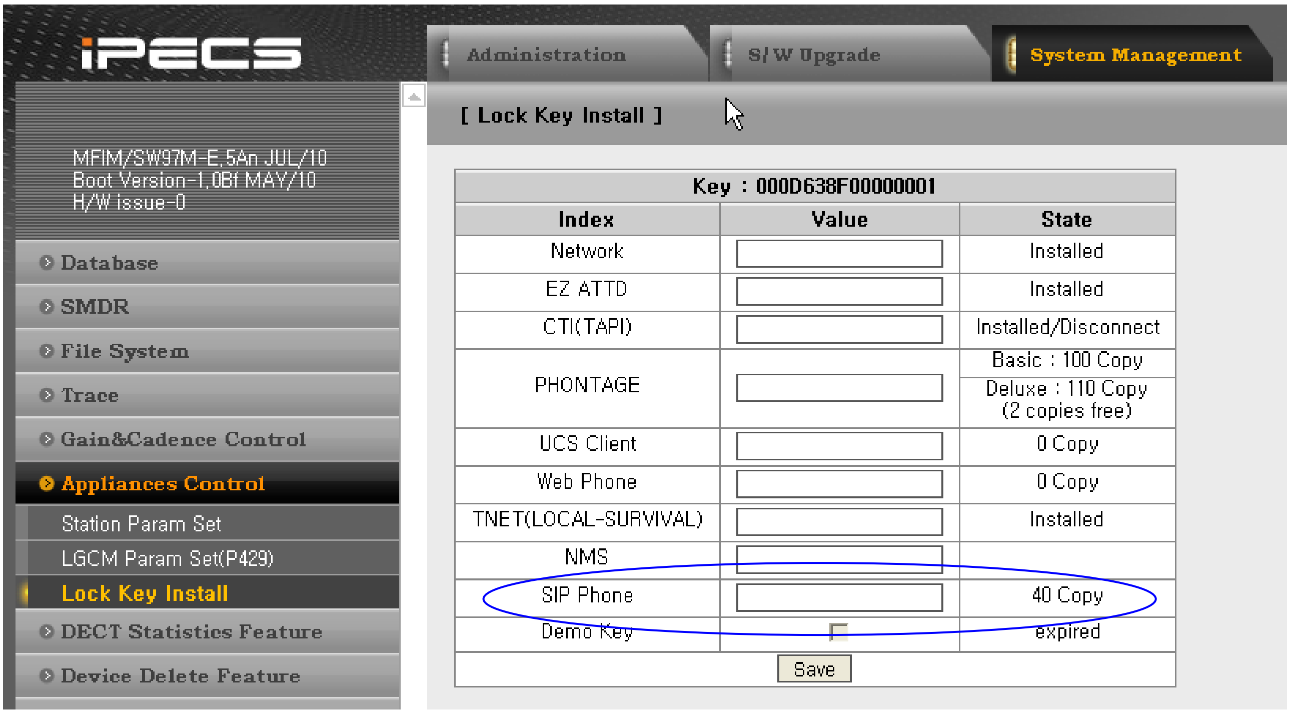
3. Input Station User Login Table(443)
   - ID (Mandatory) : An unique string or Extension Number. This should be same as SIP Phone's 'Authorization ID' value.

- Password (Optional) : A string. If you want to make authentication be implemented when user register to system, password should be available (not empty). This should be same as SIP Phone's 'Authorization Password' value.
- Desired Number (Mandatory) : Extension Number. This should be same as SIP Phone's User Name (or Phone Number) value.
Examples)

ID	Password	Zone	Desired Number
1050	dfdsfsf	1	1050
Elice	gsgsgs	1	1051
1052	1052	1	1052

If 'Save' is OK, it means that you can use the Extension number.

4. Lock Key Install
- Need not Lock Key : LG-Ericsson LIP8002, WIT400H, Yamaha ACT-50(PJP-50)
- Need Lock Key (Number of SIP Extension Copies) : Other SIP Phones except above
☞ To acquire Lock Key, contact to LG-Ericsson Enterprise Solution Sales Team.



<u>Re-Registration</u>
1. Role : a portion of keep alive / update address information of SIP Phone
2. Progress : after 1st Registration, De-Registration -> Registration
3. Re-Registration frequency : Registration Timer
4. Related Attributes

- SIP Data / SIP Phone Attributes(211) – Registration Timer Usage ON, OFF(Registration Timer by Provisioning in case of LG-Ericsson SIP Phone Registration)
- after 1st Registration, you can program related attributes in [SIP Data / SIP Phone Attributes(211) – Registration Timer Usage : ON, Registration Timer : default or more than 10 minute recommended
- or, before 1st Registration, you can set the related attributes in [SIP Data / SIP Phone Provisioning(212) for LG-Ericsson SIP Phones – rerer to SIP Phone Provisioning]

5. SIP Data / SIP Phone Attributes(211) – Registration Timer has meaning only when 'Registration Timer Usage' is 'ON'
6. Shorter timer will take more traffic on network
7. For Keep Alive usage, we recommend Keep Alive Usage ON – the 'OPTIONS' message. if a SIP Phone does not support 'OPTIONS' message then you can implement a portion of Keep Alive by this Re-Registration. In that case, set the Registration Timer to 60 seconds we recommend.

Status Monitoring (Keep Alive) and Talking Monitoring (Session Timer)

1. Status Monitoring (Keep Alive)
   - Role : When a SIP Phone is unplugged or disconnected from network, system can recognize that and make the Extension be disconnected status
   - Keep Alive Usage : SIP Data / SIP Phone Attributes(211) – Keep Alive Usage - ON
   - Keep Alive Timer : SIP Data / SIP Attributes(210) – Check Message Send Timer - 30 seconds
   - Keep Alive message Retry Count : SIP Data / SIP Phone Attributes(211) – Retry Count - 3
2. Talking Monitoring (Session Timer)
   - Role : When a SIP Phone is unplugged or disconnected from network during talking with other Phone, CO G/W or other devides, system can recognize that and disconnect the talking call to avoide locking up the associated devices and resources.
   - Session Timer Usage : SIP Data / SIP Phone Attributes(211) – Session Timer Support - ON
   - Disconnect Guard Time : SIP Data / SIP Phone Attributes(211) – MAX Session Timer - 180 second, for example.
   - Minimum Guard Time for session timer negotiation : SIP Data / SIP Phone Attributes(211) – MAX Session Timer - 60 second, for example.

Authentication for Registration and Call Setup

1. Authentication for Registration
   - Progress :
     SIP Phone (REGISTER w/o or wrong Authentication information) ---> MFIM
     SIP Phone <--- (401 unauthorized) MFIM
     SIP Phone (Registration w/ right Authentication information) ---> MFIM
     SIP Phone <--- (200OK) MFIM
2. Authentication for Call Setup
   - Progress :
     SIP Phone (INVITE w/o or wrong Authentication information) ---> MFIM
     SIP Phone <--- (407 proxy authentication required) MFIM

SIP Phone (INVITE w/ right Authentication information) ---> MFIM
SIP Phone <--- (100 Trying / 180 Ringing) MFIM

3. Related Attributes
   - SIP Data / SIP Phone Attributes(211) – 407 Authentication : ON
   - Station User Login(443) – Password – Available (not empty)

DTMF Type

1. MFIM currently support only INFO type DTMF for SIP Extension (does not INBAND and 2833).
   ☞ for SIP Trunk, all type of DTMF is supported.
2. If SIP Extension talking voice path is connected directly each other then they can implement INBAND or 2833 DTMF independently.
3. Synchronization between system and SIP Phone is required.
   - SIP Data / SIP Phone Attributes(211) – DTMF Type : One of INFO (default: DTMF RELAY)
   - SIP Phone Self Programming - DTMF Type : INFO or a specific INFO type if there is in select list.
4. By Provisioning for LG-Ericsson SIP Phone, DTMF Type of Phone is automatically set to INFO Type. In that case, you do not need to set it by SIP Phone Self Programming.

Network Related Configuration

1. MFIM IP address for WAN is different from original IP address for LAN
   - Condition 1 : MFIM has firewall IP address or MFIM is port-forwarded by VOIM WAN-U or other Switches
   - Condition 2 : There is a SIP Phone in WAN side of MFIM
   - Resolution : SIP Data / SIP Phone Attributes(211) – Same Zone with MFIM - OFF
   - Implementation : MFIM will implement signaling with its WAN IP address for the SIP Phone.
2. A SIP Phone is on NAT environment (for example, wireless SIP Phone via AP)
   - Condition 1 : The NAT IP address or IP Port of SIP Phone (WAN address of SIP Phone) is frequently updated.
   - Condition 2 : Sometimes communication is disconnected, unreachable or mismatched because of so-often updated address by NAT mapping implementation.
   - Resolution 1 : Enable the Keep Alive option for the SIP Station(s) that are on NAT environment.
   - Implementation 1: MFIM will send ‘OPTIONS’ message so often (in 30 seconds) to assist to maintain the address of SIP Phone’s WAN.
   - Resolution 2 : set static NAT address mapping by port-forwarding. For example 1 to 1 static NAT address assignment by port-forwarding in NAT switch.
   - Implementation 2: SIP Phone’s WAN address will not be changed.

SRTP

1. For Voice and Video RTP Data Encryption
2. To implement SRTP, VOIM channel is required for SRTP packet relay. You should equipped LIK VOIP G/W.
3. Usage (for example, LG-Ericsson SIP Phone)
   - SIP Data / SIP Phone Attributes(211) – SRTP Usage : ON

- SIP Phone Self Programming(Phone's Web Programming) – SRTP Usage : ON

4. Encryption Key Method (for example, LG-Ericsson SIP Phone)
   - SIP Data / SIP Phone Attributes(211) – 1st CRYPTO : select one of listed
   - SIP Data / SIP Phone Attributes(211) – 2nd CRYPTO : select one of listed
   - SIP Phone Self Programming(Phone's Web Programming) – 1st CRYPTO : select one of listed
   - SIP Phone Self Programming(Phone's Web Programming) – 2nd CRYPTO : select one of listed if the SIP Phone support 2nd CRYPTO option
5. For other 3rd party SIP Phone, you need to install SRTP Encryption Method on that Phone, at first. More technical information may be required for you.

Other SIP Phone Attributes

1. SIP Data / SIP Phone Attributes(211) – Registration Mode
   - by default, Register : Re-Registration should be implemented from the SIP Phone
     ☞ MFIM will set the Re-Registration Timer for the SIP Phone. If there is no Re-Registration from the SIP Phone during the Registration Timer then system will disconnect the SIP Phone automatically after Registration Timer expires.
   - Manual : Re-Registration is not required for the SIP Phone
2. SIP Data / SIP Phone Attributes(211) – TRANSPORT
   - by default, UDP
   - TLS : you need to set the Encryption Key. More technical information may be required for you.
3. SIP Data / SIP Phone Attributes(211) – SIP Phone Type
   - Automatically determined by system
4. SIP Data / SIP Phone Attributes(211) – Device NAT Usage
   - AUTO : system determine automatically

Multi Line support for a SIP Phone

1. LIK allow multiple Extension for a SIP Phone that support Multi Line.
2. A SIP Phone will implement multiple register to system
3. System will distinguish them by,
   - 1st Extension : write it as a device with original MAC address
   - from 2nd Extension : write it as a device with virtual MAC address

Conditions

4. You need to install Lock-Key for SIP Phones except LG-Ericsson LIP8002, WIT400H, Yamaha ACT-50(PJP-50).

Programming

SIP DATA

1. SIP Attributes (210)
2. SIP Phone Attributes (211)
3. SIP Phone Provisioning (212)

Related Features

Hardware

MFIM1200:

Support up to 600 channels (SIP extension + CO trunk) simultaneously

MFIM600:

Issue 0, 1, 2: Support up to 100 channels (SIP extension + CO trunk) simultaneously

Issue 3 : Support up to 200 channels (SIP extension + CO trunk) simultaneously

MFIM300:

Support up to 200 channels (SIP extension + CO trunk) simultaneously

MFIM100:

Support up to 100 channels (SIP extension + CO trunk) simultaneously

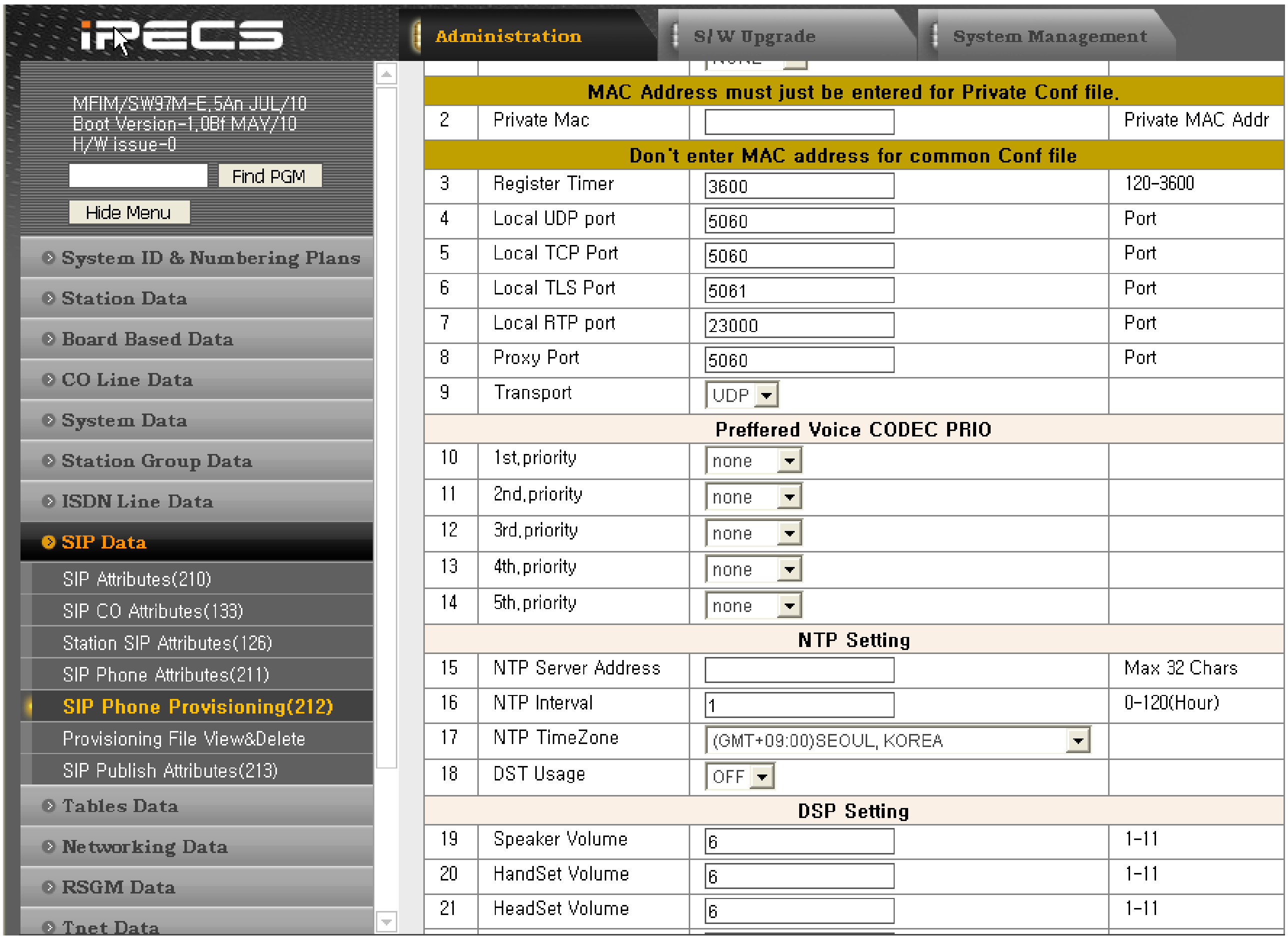
iPECS-50A/B, Micro:

Support SIP extension and CO trunk according to the maximum station and CO lines.

8.2 Provisioning

Description

By default, you do not need to implement provisioning. Provisioning is only for LG-Ericsson SIP Phone types. If you do not want to implement provisioning, you can set the same attributes in SIP Data / SIP Phone Attributes(211) per a SIP Extension or by range.



< Why ? >

To pre-assign default attributes and download configuration to SIP Extensions when they register to System (MFIM)

< For Who ? (for all of specified Phone type or for one MAC specified Phone) >

1. CONFTYPE : select Phone Type / Mandatory

< LG-Ericsson WIT400H >

- Currently MFIM (tftp only) does not proceed provisioning for WIT400H(http only) because of different method

☞ But, WIT400H follows LIK system's default provisioning by itself automatically.

< LG-Ericsson LIP8002 / LIP88xx >

- MFIM proceed provisioning for LIP8002 / LIP88xx

< Other 3rd party SIP Extensions >

- Does not proceed provisioning

2. Private Mac : specify MAC address if provisioning target is only for one specific SIP extension / Optional

< For What ? >

1. Re-Registration Timer
   : this will be useless if [SIP Phone Attributes (PGM 126) - Registration Timer Usage] is ON
2. SIP Extension’ s Local UDP/TCP/TLS Port number
3. Proxy Port : Server port number in sight of SIP Extension toward MFIM
4. Transport : Signaling mode
5. SIP Extension’ s CODEC Priority
6. NTP Server and DST setting
7. default volume of Speaker/Handset/Headset, maximum volume of Handset
8. Digit Map
   If you can utilize SIP Phone’s programmable Dial Digit Map, you can make immediate sending out dials w/o press [SEND] or ‘#’ key.
   <Refer to below>
   - Digit Map (for Call Setup in SIP Phone)
     : you can program Digit Map according to System Numbering Plan
     Example) in case of ‘LIP8002/88xx’ SIP Phone type
     x.T|0|xx#|xxx#|1xxx|2xxx|3xxx|xxxx+#
     Explanation)
     ‘9’ : dial ‘9’ for attendant call, you can specify other single digit if you want it to be sent immediately when press the digit without ending by ‘#’
     ‘1xxx’, ‘2xxx’, ‘3xxx’ : extension numbering, you can specify other 2 or 3 or 4 or more digits starting with a specific prefix digit or digits
     ‘xx#’, ‘xxx#’, ‘xxxx#’ : dial the other 2, 3 or 4 digits and press ‘#’ to send
     ‘xxxx + #’ : dial the other more than 4 digits and press ‘#’ to send
   - Digit Map (for 2nd Dial Tone) : Play self dial tone by SIP Phone
     Example) in case of ‘LIP8002/88xx’ SIP Phone type
     0 // ‘0’ is CO access code in the first CO group

Operation

SIP Data / SIP Phone Provisioning(212)

1. Select one of CONFTYPE (LG-Ericsson SIP Phone Type)
2. By default you do not need to input Private MAC address because you want to make provisioning for all of CONFTYPE SIP Phones. If you want to make provisioning for a specific individual SIP Phone then input Private Mac Address of the SIP Phone.
3. Set the attributes you want to set for those type of SIP Phones on their registration to system and press save button.
4. To view the saved information, press the view button.
   ☞ Obnormal Case : If you cannot see the saved provisioning data even though you saved them, then you can try to solve the problem as below.
   Cause of Problem : MFIM does not have enough working memory space, currently.
   Resolution : Shut down ‘SIP Server’, save Provisioning, Start ‘SIP Server’
   Step 1) Shut down SIP Server : connect COM1 and enter into maintenance mode, and type

maint> sipstack kill
// 'SIP stack:kill(kill/start/refresh)' will be displayed on the screen

Step 2) Go to [SIP Data / SIP Phone Provisioning(212)] and implement saving Provisioning

Step 3) Start Up SIP Server : connect COM1 and enter into maintenance mode, and type
maint> sipstack start
// 'SIP stack:start(kill/start/refresh)' will be displayed

ATTRIBUTE	DESCRIPTION	RANGE	DEFAULT
CONFTYPE	Mandatory, select one of LG-Ericsson SIP Phone type		
Private MAC	Specify MAC address of a SIP Phone to serve private provisioning for a Extension. For common provisioning for all of CONFTYPE SIP Extensions, do not specify MAC address		
Registration Timer	Re-Registration Timer		
Local UDP Port	SIP Phone default signaling UDP port		
Local TCP Port	SIP Phone default signaling TCP port		
Local TLS Port	SIP Phone default signaling TLS port		
Local RTP Port	SIP Phone default RTP port start range		
Proxy Port	SIP Server port for SIP Phones (MFIM SIP Port number)		
Transport	Default signaling method		
CODEC			
1^{st}. priority ~ 5^{th}. priority	CODEC priority		
NTP Setting (Need for TLS)			
NTP Server Address	NTP server IP address.		
NTP Interval	Interval		
NTP Time Zone	Time Zone		
DST Usage	Daylight Saving Time		
DSP Setting			
Speaker Volume	Default volume level of SIP Phone		
Handset Volume	Default volume level of SIP Phone		
Headset Volume	Default volume level of SIP Phone		
MAX Handset Volume	Default volume level of SIP Phone		
Digit Map			
Dial Tone Digit	Second Dial Tone digit specification in SIP Phone		
Pause Timer	Dial Pause timer in SIP Phone		
Digit Map	Send setup to system numbering plan		
Emergency Code	Emergency Codes of system. With this emergency code Phone will override lock of making call		
System Setting			
Feature Sync.	ON : Do Not Disturb and Call Forward feature synchronization with system when SIP Phone set the call feature in phone side.		ON
Auto Idle Timer	Duration of call-ended status till go to idle		
Save : save provisioning for the specified common CONFTYPE or specific SIP Extension with Private MAC View : display all of saved provisioning information Cert : extract cert data to system if there is any uploaded cert key data and display the information			

Conditions

1. Provisioning is only for LG-Ericsson SIP Phone types.

Programming

SIP DATA

1. SIP Phone Attributes (211)
2. SIP Phone Provisioning (212)

Related Features

Hardware

8.3 Make Call

Description

There are three types of call setup – Station Call, CO Call, System Call Feature Implementation.

Operation

Making Station Call Setup

1) Dial Station Number
2) If 'Dial Digit Map' is programmed, SIP Phone will send call setup immediately.
3) If 'Dial Digit Map' is not programmed, press the 'SEND' button or '#' key for send out call setup.

Making CO Call Setup

1) CO Access Code + Dial Number + [SEND]
   ex) CO Access Code '0', Dial Number '450-4500',
   dial '04504500' and press [SEND] or '#' button
   ☞If you program 'Second Dial Tone Digit Map' on SIP Phone, you will hear self dial tone from SIP Phone.
2) CO Access Code + [SEND],
   after hearing CO dial tone, press Dial Number
   ex) CO Access Code '0', Dial Number '450-4500',
   dial '0 and press [SEND] or '#'
   hear CO dial tone from system
   dial '4504500'

Making System Call Feature Setup

1) System Call Feature by Numbering Code : System Numbering Plan (PGM106-109)
2) Enblock Dialing : System Call Feature numbering code + data + [SEND]
3) Supported Call Features by Numbering are,
   Internal Page Zones
   Internal All Call Page
   Meet Me Page
   External Page Zone
   External All Call Page
   All Call Page
   SMDR Account Code Enter
   SLT Last Number Redial
   Do-Not-Disturb(DND)
   Call Forward
   Speed Dial Program
   SLT Speed Dial Access
   DND/FWD Cancel
   Alarm Reset
   Group Call Pick-Up
   Universal Night Answer
   Account Code With Bin
   Walking COS
   Call Park Locations
   Station Group Pilot Number
   Station User VSF Features Access
   Call Coverage Ring

Direct Call Pick-Up
Access CO Group
Access Individual CO/IP
Access Held Individual CO/IP
Access CO In First CO Group
Attendant Call
Door Open
ENTER INTO CONF-ROOM
ENTER INTO CONF-GROUP
STATION ICR
PICKUP GROUP PICK-UP
EMERGENCY PAGE

Conditions

1. If you can utilize SIP Phone's programmable Dial Digit Map, you can make immediate sending out dials w/o press [SEND] or '#' key.
<Refer to below>

- Digit Map (for Call Setup in SIP Phone)
: you can program Digit Map according to System Numbering Plan
Example) in case of 'LIP8002/88xx' SIP Phone type
x.T|0|xx#|xxx#|1xxx|2xxx|3xxx|xxxx+#
Explanation)
'9' : dial '9' for attendant call, you can specify other single digit if you want it to be sent immediately when press the digit without ending by '#'
'1xxx', '2xxx', '3xxx' : extension numbering, you can specify other 2 or 3 or 4 or more digits starting with a specific prefix digit or digits
'xx#', 'xxx#', 'xxxx#' : dial the other 2, 3 or 4 digits and press '#' to send
'xxxx + #' : dial the other more than 4 digits and press '#' to send
- Digit Map (for 2nd Dial Tone) : Play self dial tone by SIP Phone
Example) in case of 'LIP8002/88xx' SIP Phone type
0 // '0' is CO access code in the first CO group

Programming

Related Features

Hardware

8.4 CALL ANSWER

Description
A SIP Phone will accept a call when it receives call on idle state.
On busy state, it is different according to 'Call Wait' supported or not.

Operation
Receives 1st call on Idle State

1) Accept the call and respond with Ringing
   Pickup Handset to answer the call

Receives 2nd call on Busy State

1) If do not support Call Wait or disabled
   Reject the call by '486' busy
2) If support Call Wait and enabled
   Accept the call and respond with Ringing
   The call is waited on the SIP Phone
   SIP Phone will serve self indication for the second call.

Handsfree Automatic Answer

1) By SIP Phone : self programmable option, always
2) By System : in case of receiving call by Voice Over, Intrusion, Forced Handsfree, Paging
   - LG-Ericsson SIP Phone only supported

Conditions

Programming

Related Features

Hardware

8.5 Call Hold

Description

SIP Phone make a call on hold. The other party goes to held state and hear Music On Hold(MOH).

Operation

Make a Call On Hold

1) Press HOLD button
2) or Press TRANS button and OnHook
3) Call is automatically exclusively held on system
4) Only the Extension who made the call held can retrieve the call

Music On Hold

1) System serves MOH from MISC to SIP Phone via VOIP/VOIM channel.

Held Recall

1) After programmed recall timer, system makes a call for recall to the SIP Extension who hold the call originally.

Conditions

1. To serve MOH for SIP Phone, VOIP/VOIM channel for RTP Packet Relay is required.

Programming

Related Features

Hardware

8.6 Call Transfer

Description

SIP Phone supports Consultation Transfer and Blind Transfer. Consultation Transfer is also separated to Screened and Unscreened Transfer by System

Operation

Consultation Transfer

1) During talking on a call, press [TRANSFER] button
2) Dial destination number
3) Destination is ringing and answers the call – Screened
   Or, Destination is ringing – Unscreened
4) Press Transfer Complete button or OnHook

Blind Transfer

1) During talking on a call, press [BLIND TRANSFER] button
2) Dial destination number
3) Press [SEND] or Call Complete button
4) SIP Phone will goes to idle automatically.

Transfer Recall

1) When call transfer failed or no answer, there will be recall to the transferor SIP Phone.

Conditions

Programming

Related Features

Hardware

8.7 CALL FORWARD

Description

LIK System support Call Forward by SIP Phone. SIP Phone also can utilize Call Forward Feature of System.

Operation

Call Forward Authority ON

1) System Admin – Station Data – Station Attributes(111-113) – Call Forward : ON

Call Forward setting by SIP Phone

1) 3rd Party SIP Phone
   - Self Programming
2) LG-Ericsson SIP Phone
   - Self Programming w/o Provisioning or w/ Provisioning in which 'Feature Sync' is OFF.

Call Forward setting by System

1) 3rd Party SIP Phone
   - Dial 'Call Forward' Feature Code + Condition + Destination + [SEND]
2) LG-Ericsson SIP Phone
   - Self Programming w/ Provisioning in which 'Feature Sync' is ON.
   ☞ With Feature Sync ON, Call Forward data by Self Programming will be notified to system and applied to system's station database. By this proprietary implementation, system can recognize Call Forward and DND setting of SIP Phone side. Also, the real Call Forward and DND oprtation is implemented by system.
   - Or, Dial 'Call Forward' Feature Code + Condition + Destination + [SEND]

One Time Call Forward for a Ringing Call

1) A call is ringing to a SIP Phone
2) SIP Phone user press Call Forward button and dial destination number
3) The ringing call is forwarded to dialed destination

Conditions

1. To follow up call forward for SIP Phone, call forward authority should be ON in Station Programming(111).

Programming

Related Features

Hardware

8.8 Do Not Disturb (DND)

Description

LIK System support Do Not Disturb (DND) by SIP Phone. SIP Phone also can utilize Do Not Disturb Feature of System.

Operation

DND setting Authority ON

1) System Admin – Station Data – Station Attributes(111-113) – DND : ON

DND by SIP Phone

1) 3rd Party SIP Phone
   - Self Programming
2) LG-Ericsson SIP Phone
   - Self Programming w/o Provisioning or w/ Provisioning in which 'Feature Sync' is OFF.

DND setting by System

1) 3rd Party SIP Phone
   - Dial 'DND' Feature Code + [SEND]
2) LG-Ericsson SIP Phone
   - Self Programming w/ Provisioning in which 'Feature Sync' is ON.
     ☞ With Feature Sync ON, DND setting by Self Programming will be notified to system and applied to system's station database. By this proprietary implementation, system can recognize DND setting of SIP Phone side. Also, the real DND oprtation is implemented by system.
   - Or, Dial 'DND' Feature Code + [SEND]

One Time DND for a Ringing Call

1) A call is ringing to a SIP Phone
2) SIP Phone user press DND button
3) The ringing call is rejected or only ring is muted according to SIP Phone's implementation

Conditions

1. To follow up DND for SIP Phone, DND authority should be ON in Station Programming(111).

Programming

Related Features

Hardware

8.9 CONFERENCE

Description

SIP Phone who has 3-way conference capability can make conference call by Phone itself without utilize system's conference feature. Also, SIP Phone can utilize system's conference feature – Conference Room and Conference Group. To serve system conference for SIP Phone, a mixing device – MCIM is required.

Operation

SIP Phone Self Conference

1) Make a call and connected
2) Press 'Conference' button
3) Dial 2nd call and connected
4) Press 'JOIN' button

Or,

1) Make a call and connected
2) Press 'HOLD' button
3) Dial 2nd call and connected
4) Press '3-way Conference' button

System Conference Feature

1) Conference Room : refer to 2.23.1 Conference Room – SLT Operation
2) Conference Group : refer to 2.23.4 Conference Group – Operation
3) System need to be equipped MCIM module to mix voice

Conditions

1. To serve system conference for SIP Phone, a mixing device – MCIM is required

Programming

Related Features

Hardware

8.10 CALL WAIT & BROKER CALL

Description

When SIP Phone receives second call wait, user can switch talking to each other. There are a few kinds of call wait and according to the type of call wait operation of switching talk is different.

Operation

Receive 2nd call

1) In Case of, SIP Phone has no Call Wait feature or Call Wait is disabled
   - If caller is from CO line then the call will be rerouted by system setting automatically
   - If caller is from a Station then the caller will hear busy tone. During busy tone, the user can press '*' for Camp-On call or '#' for Voice-Over. The SIP Phone user who receives this 2nd call will hear notification tone from system.
   - The SIP Phone user who receives this 2nd can make hold for 1st call talking and switch to 2nd call for talking by press 'HOLD' twice

2) In Case of, SIP Phone has Call Wait feature and Call Wait is enabled
   - The 2nd call is accepted by the SIP Phone and user who receives this 2nd call will hear notification tone from Phone itself.
   - The caller from CO or Station will hear ringback tone ore coloring.
   - The SIP Phone user who receives this 2nd can make hold for 1st call talking and switch to 2nd call for talking by,
     a) 3rd party SIP Phone : press 'HOLD' twice
     b) LG-Ericsson WIT400H SIP Phone : press 'HOLD' once
     c) LG-Ericsson LIP88xx and LIP8002 SIP Phone : press Up or Down of Navigation button

Hold 1st call and Make 2nd call

1) SIP Phone user makes hold with 1st call and makes 2nd call.
2) After 2nd call is connected the SIP Phone user talks with 2nd call party.
3) If he want to return to 1st call party without hangup the 2nd call,
   - 3rd party SIP Phone : press 'HOLD' once
   - LG-Ericsson WIT400H SIP Phone : press 'HOLD' once
   - LG-Ericsson LIP88xx and LIP8002 SIP Phone : press Up or Down of Navigation button

Conditions

1. Broker Call (Switch of Talking) operation maybe different according to what kind of SIP Phone type.

Programming

Related Features

Hardware

8.11 CALL PARK

Description

SIP Phone user can park a talking call to system's Call Park Location. And the other Station user can retrieve the parked call. The parking operation is 'Blind Transfer'.

Operation

Call Parking

1) SIP Phone user is on a CO talking.
2) SIP Phone user park the talking call and the call will be parked to system's Call Park Location
   - press 'Blind Transfer' button
   - dial Park Bin number
   - [SEND]
3) The SIP Phone user will page to inform somebody to pickup the call.
4) Somebody will retrieve the parked call.

Conditions

1. Transfer for Call Parking should be implemented by Blind Transfer operation.

Programming

CALL PARK LOCATIONS	1.	Numbering Plan (PGM 106 ~ 109) – Call Park Locations

Related Features

Hardware

8.12 MCIM REQUIREMENT FOR SIP PHONE

Description

For SIP Phone, system should be equipped MCIM module to serve below features.

1. Conference Room
2. Conference Group
3. Voice Over
4. Intrusion
5. Automatic Two-Way Recording

Conditions

Programming

Related Features

Hardware

8.13 VOIP or VOIM8/24 Requirement for SIP Phone

Description

For SIP Phone, system should be equipped VOIP or VOIM8/24 module to serve below features.

1. CO Line for H323, Networking, SIP Trunking
2. DSP for generation of Busy/ Error/ Confirm/ Ring-Back/ Hold/ Page/ Warning/ OHVA/ Intrusion/ Dial tones from system to SIP Phone
3. Relay of Music On Hold from system to SIP Phone
4. Relay of Paging from/to SIP Phone
5. Voice RTP Packet Relay between private LAN and public WAN, local and remote, NAT resolution

Conditions

Programming

Related Features

Hardware

8.14 CALL TRACE OF SIP EXTENSION FOR DEBUGGING

Description

If debugging is required, you can set call trace on MFIM for SIP Extensions.

Operation

Basic

t s call
t s rawdata
t b [slot#]

add/remove SIP Extension Call Trace

t s sipext // add
t r sipext // remove

add/remove SIP Stack Trace

t s sip // add
t r sip // remove

add/remove SIP Packet Trace

t s fsipm // add
t r fsipm // remove

remove all of Trace

t r all // remove all type of trace
t d // remove all of devices

Trace Direction

p v // to view trace direction
p s // to set trace direction

Conditions

Programming

Related Features

Hardware

8.15 DSS/BLF (Busy Lamp Field)

Description

BLF feature is supported with LG SIP Phone (SIP-88xx).

Operation

To set DSS/BLF feature in SIP-88xx

6. Enter the SIP-88xx Web administration
7. Enter the Button Assignment menu.
8. Select DSS/BLF and set the BLF number.
9. Press [Change] button.

To call directly using DSS/BLF button

1. Press the DSS/BLF(1003) button that you call to directly
2. The LED of button(1003) will blink.
3. If station 1003 answers the call, the LED of DSS/BLF will be on.

To pick up call forDSS/BLF button

1. The LEDd of DSS/BLF(1003) will blink while station 1003 is receiving call
2. By pressing DSS/BLF(1003), call will be picked up..

Conditions

4. DSS/BLF can just show 3 status – Idle(LED 'off"), Ringing(LED 'blink') and Busy(LED 'on')
5. It just show the status related with call – For DND and other features the LED will be off.

Programming

Related Features

Hardware

None

8.16 SIP VIDEO CALL

Description

Video Call feature is added

Operation

To place Video call from SIP-88xx

1. Dial the counterpart Video phone number and press [Video Call] button. Or press [Video C all] button and dial the counterpart Video phone number.
2. If the counter part can answer with Video, Video call will be established. If not, voice call wil l be established.

To establish Video call after answering with voice call

1. Voice call will be established at first.
2. Start Video codec negotiation pressing [Video Start] button or soft menu
3. Video call will be established if the video codec negotiation is succeeded.

Conditions

6. After pressing [CONF], [HOLD], [TRANS], Video call will be changed as Voice call.
7. For the Trunk outgoing call, can not establish Video call directly – after establishing Voice call, Video call must be established.

Programming

Related Features

Hardware

None

8.17 SIP VIRTUAL MOBILE EXTENSION (VMEX)

Description

SIP gives us the opportunity to provide a truly integrated and cost effective mobile extension solution. The closer integration lets the mobile users to use the mobile exactly as they are used to.

The goal have been to integrate the mobile phone even further with the PBX (out-band signaling) and add a more cost effective solution for bigger systems (recursing diversion). The out-band signaling is mandatory. Recursing diversion is not required in smaller solutions, but still more cost effective. In a bigger solution, more then 20-30 MEX users, it is needed to compete with Aastra, Cisco and various hosted solutions, including the hosted solution we sell.

Out-band signaling makes it a truly transparent solution. The user places an outgoing call exactly as he is used to and it works on any cell phone. This means that the user can use speed dials and do not need to have any special phone or application installed in the phone. Also the use of DISA, which is considered too complicated by many users, will be obsolete.

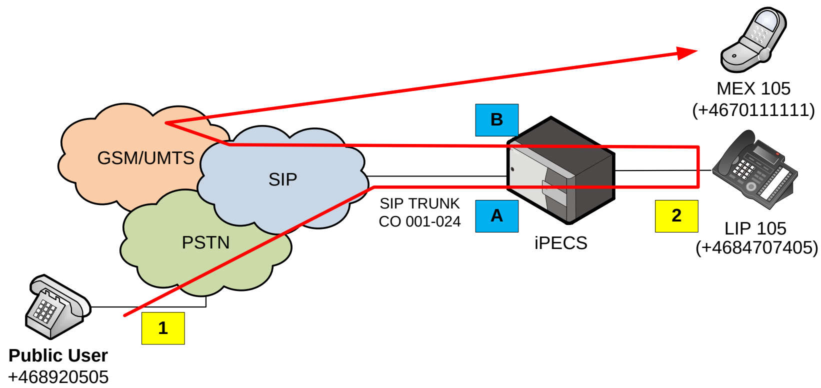
Operation

1. Incoming call (from PSTN to MEX)

TDC uses a prefix on the To tag and From/Contact tag. These prefixes are trunk based and should be added to the mobile extension section of pgm 133.

1. The Public user calls 084707405 (LIP 105) and an Invite [A] is sent to iPECS.
2. LIP 105 has MEX and the call should be routed to MEX 105.

   We now need to add two prefixes to the outgoing Invite [B] and they are described below.



<u>[A] Incoming Invite</u>

From: <sip:+468920505@se-pgw1.ipt.ip.tele.dk:5060;user=phone>;tag=1399855756

To: <sip:+4684707405@btse.ip.tdk.dk:5060;user=phone>

Contact: <sip:+468920505@62.242.160.7:5060;transport=udp>

<u>[B] Outgoing Invite</u>

To: <sip:+**46394980**4670111111@tdc.se>

From: <sip:+**46752468004**468920505@tdc.se>;tag=61D3F1B4-1ACC

Contact: <sip:+**46752468004**468920505@195.84.135.11:5060>

- The incoming and the outgoing Invite looks as they do today except for two prefix that are added to the outgoing invite.

- Prefix 1 (Red) is always added to the To header. This prefix is fixed and used by all customers.
- Prefix 2 (Blue) is always added to the From and Contact header. This prefix is a trunk ID and each trunk has a unique prefix.

Example of new admin programming

The prefixes are only used in case of mobile extension and should therefore be related to the MEX programming in pgm 133.

Mobile Extension Call	Range	Default
From/Contact ID	EXT ORG Fixed Table	EXT
P-Asserted-Identity		EXT
Remote-Party-ID		EXT
Diversion	Not Use/EXT/ORG/Fixed	Not Use
To Prefix	24 characters	
From/Contact Prefix	24 characters	

2. Outgoing call (from MEX to internal station or external number)

Outgoing call requires that iPECS can detect out-band signaling and analyze that number to decide if it is an external call, internal call or some other command.

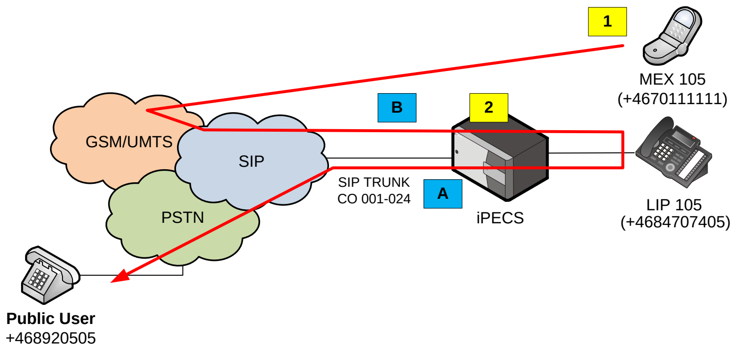
1. MEX 105 calls the Public user 08920533 and an Invite [B9 is sent to iPECS.

2. iPECS detects that it is a MEX call by finding the prefix 9999 in the To header.

   iPECS checks if the number in the From header exists in MEX table (pgm 236).

   iPECS checks the number after 9999 and detects that it is an external number.

   iPECS sends an outgoing Invite [A] to the Public user.



[B] Incoming Invite

To: <sip:+**9999**08920533@tdc.se>

From: <sip:70111111@se-pgw1.ipt.ip.tele.dk:5060;user=phone>

9999 is a prefix with 3 to 5 digits that can be chosen freely by the customer. When a call is placed from the mobile phone the To header format is 9999 + dialed number. It does not matter what kind of number the mobile user dials.

[A] Outgoing Invite

Normal Invite with the same format as if it was LIP 105 who made a call to the Public user.

Analyzing the called party number

When we have removed 9999 from the To header we have to detect what kind of number the remaining digits are. Is it an internal number (station or group number), feature code or external number? With feature code I mean the codes used for "Remote control for Mobile Extension in iPECS" described in SE-LIC-08056. The calling party number could be check like this;

1. Is it a station/group number? If YES; setup a call to that station/group. If NO; continue.
2. Is it a feature code? If YES; enable/disable that feature. If NO; continue.
3. If none of the above it is an external number. Setup an external call.

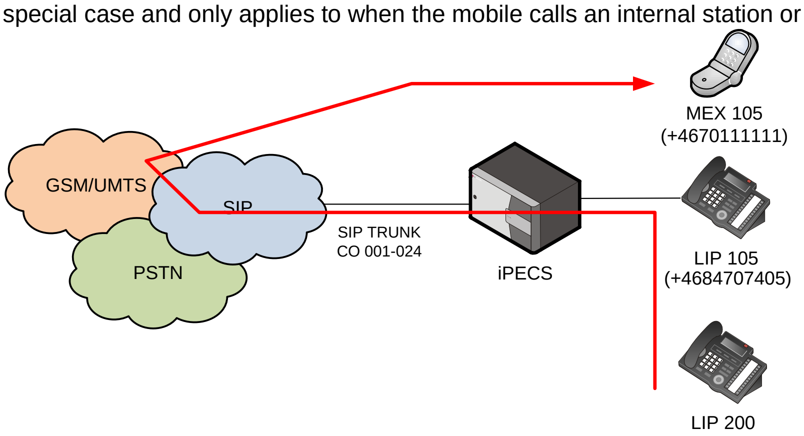
Example of new admin programming

Pgm 133: SIP MEX Out-band Prefix: 1 to 8 digits (to define the out-band prefix)

Pgm 133: Use MEX Out-band Signaling: Yes/No

3. Displaying internal station number

If we display internal station numbers for internal calls the feature feels more integrated and gets easier to use. This is a special case and only applies to when the mobile calls an internal station or when an internal station calls the mobile.



- When LIP 200 calls MEX 105 it should display 200.

- When MEX 105 calls LIP 200 it should display 105.

Conditions

1. There is no linked hard phone in system.
2. Need SIP Extension Copyright

Programming

1. Registration

- System ID & Numbering Plans / PGM 101

  'Virtual Registration'

  Device ID : STA / VMEX SIP

  MAC Address : automatic virtual assigned

  Data : Station Number / Replace

  - create new VMEX

    Station Number = a new or unused station number

    Replace = uncheck

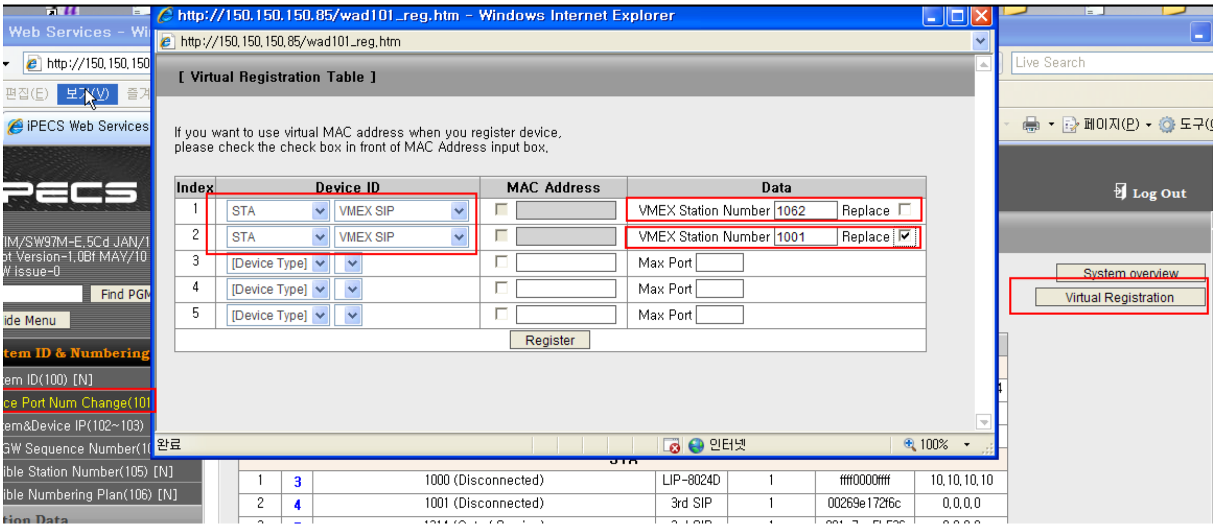
  - replace non-VMEX station to VMEX type

    Station Number = a previous used station number should be in disconnected or 'Out Of Service' state.

    (you can force a station 'Out Of Service' state in PGM 103

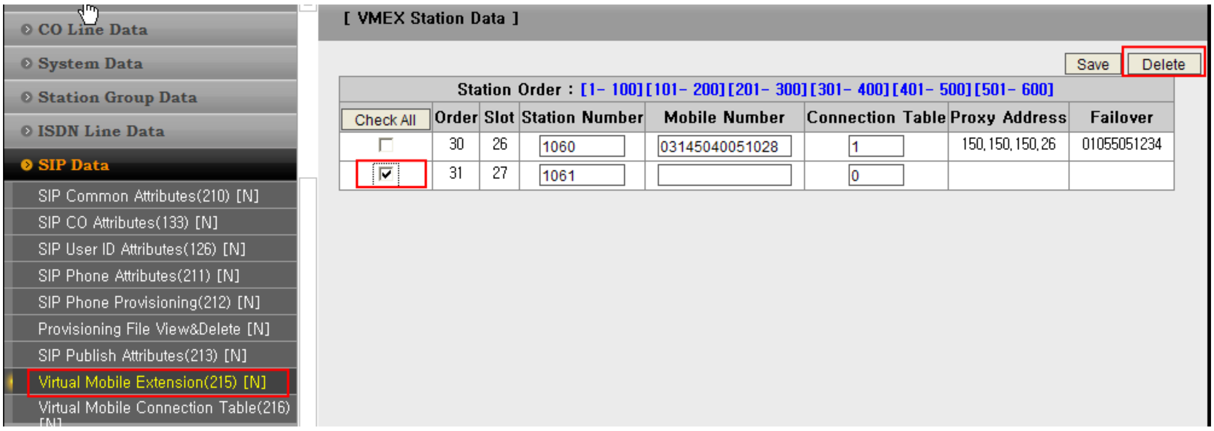
    : click 'STA', check 'SVC' and save)

    Replace = check



2. Unregister VMEX

- SIP Data / PGM 215
- check a station or stations and implement 'delete' button operation
- When a station is unregistered from VMEX it goes to 'Out Of Service' state automatically.
  To activate the station for other user, you can replace type and MAC and force 'Service' state by click 'STA', check 'SVC' and save in PGM103.
  Also, you can replace that station automatically by station login with ID/PWD table in PGM 443.
  In PGM 443, just input same station number as the deactivated station and that will be activated automatically.



3. VMEX attributes

- SIP Data / PGM 215
  1) Station Number : you can change station number and this is same as you do in PGM 105
  2) Mobile Number
     - a incoming CLI from remote Mobile Phone : address of SIP 'From' message header
     - This is ID of SIP Extension.
       With this CLI the Mobile is identified as an extension
  3) Connection Table
     - Outgoing destination address to Mobile Phone
     - Connection IP information of Mobile Phone

- index of connection table

4) Failover

- Failover information
- PGM 236 Mobile Extension Table

  Usage : Fail Over

  CO Group : a public CO

  Telephone Number : a public telephone number

c.f) currently, I have a thing to do more.

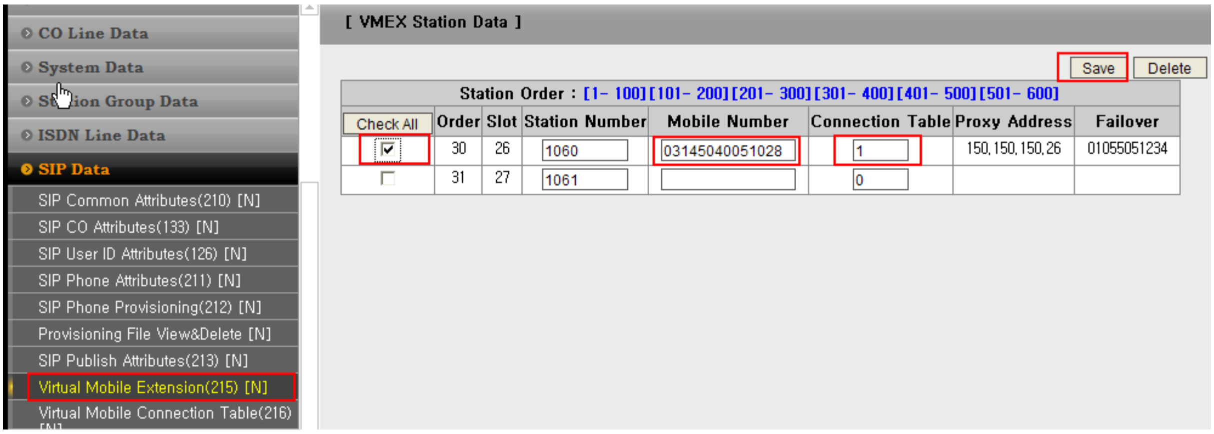
In this version, failover is implemented when receive '408 time out' message from SIP stack.

You should wait somewhat long time to meet '408 time out' message after dial the VMEX.

And there is no ringback tone when failover is tried.

I will add failover time in PGM 211 per stations and will modify for ringback tone serve.

This failover is not only for VMEX but also for all of SIP extensions.



4. VMEX Connection Table

- SIP Data / PGM 216
- Number of table is 10 because there can be some of different trunk over different SIP service provider.
- Proxy IP : IP address of VMEX over SIP service provider
- Domain Name : domain name instead of IP address will be supported later.

  currently you can input same IP address as Proxy IP.
- Port : SIP Port Number of VMEX over SIP service provider
- Firewall : Apply = use firewall IP of MFIM ; Not = use LAN IP of MFIM
- E164 type : It was originally modification of incoming invite from Mobile by removing nation and area code manually.

  But currently it is just for reserved because incoming invite is automatically modified.

  for example, if To: <sip:+999908920533@tdc.se> then +999908920533 is converted to '999908920533' and

  if the '9999' is incoming prefix then it is converted again to '08920533' and if '089' is nation and area code

  then the last invite will be '20533'.

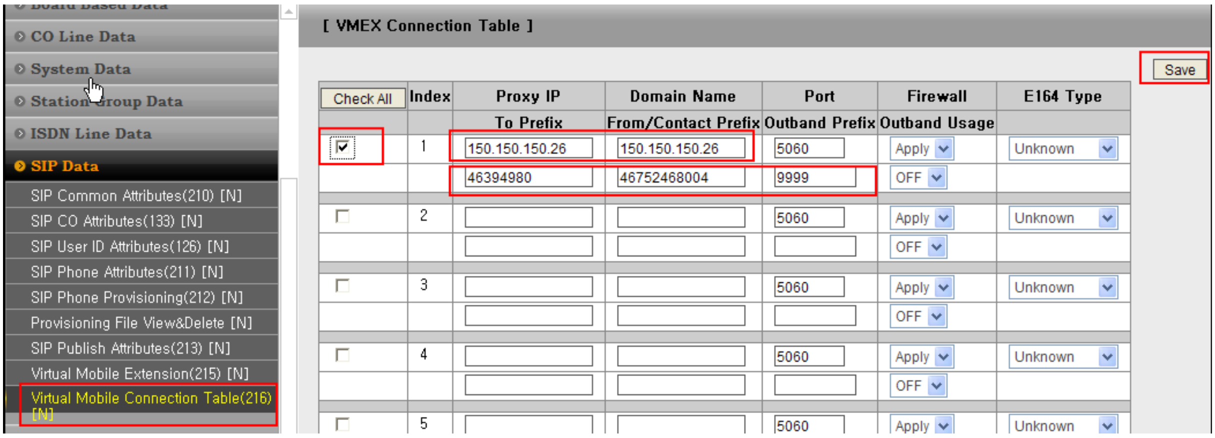
  c.f) If you need to modify manually by E164 type, I will change to see this attribute.
- To Prefix : add this in To:header when invite to Mobile Phone
- From/Contact Prefix : add this in From: and Contact: header when invite to Mobile Phone
- Outband Prefix : analize this for To: header of invite from Mobile Phone.

Remove this prefix if it is matched.

Reject the invite if this prefix if it is not matched.

c.f) If you mean this as an authentication number per VMEX then I should move this to PGM 215, per VMEX.

- Outband Usage : currently reserved and it will be for authorization ON/OFF of incoming invite from Mobile Phone.



5. RTP Connection

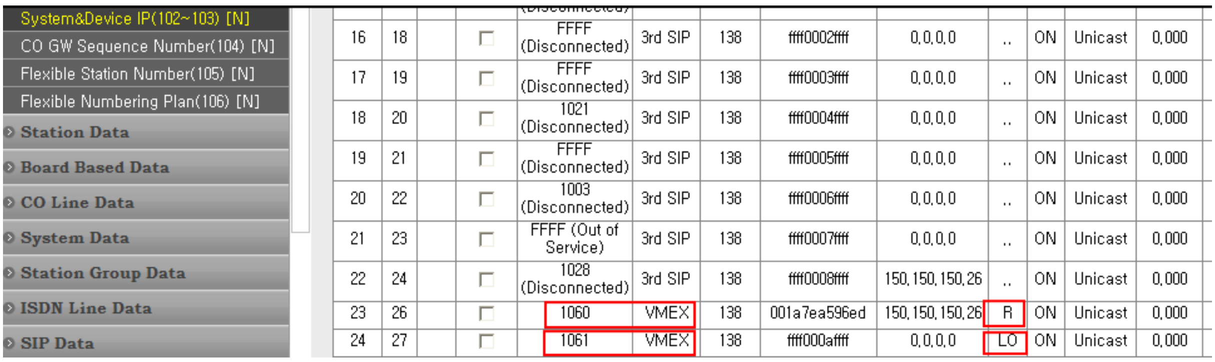
- VMEX is registered as a 'remote' mode phone by default.

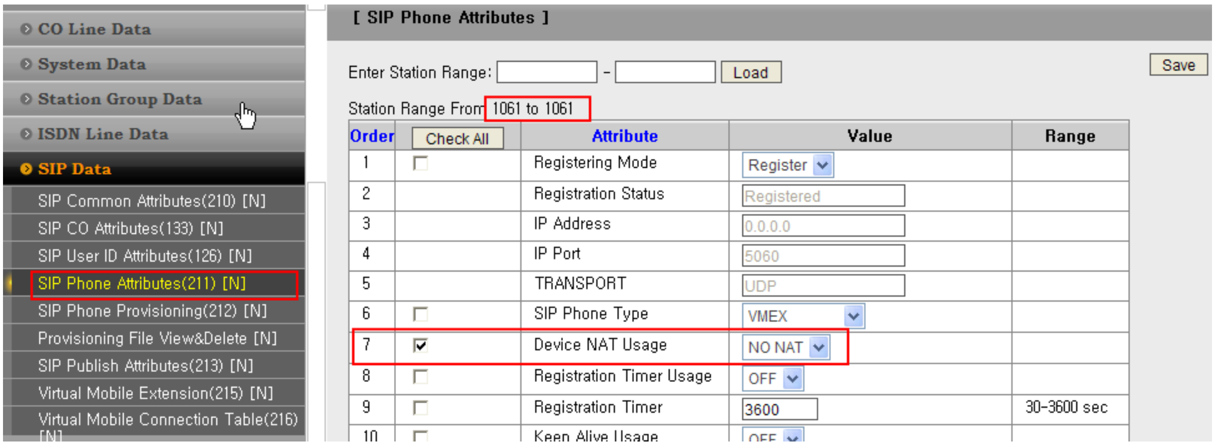
And there need one VOIP relay channel between VMEX and internal local mode station, CO, VSF/VMIM or misc device.

If you want to force VMEX to local mode, you can make that by set PGM 211 - Device NAT Usage = 'NOT NAT'.

If then you need to establish all of routable IP configuration.

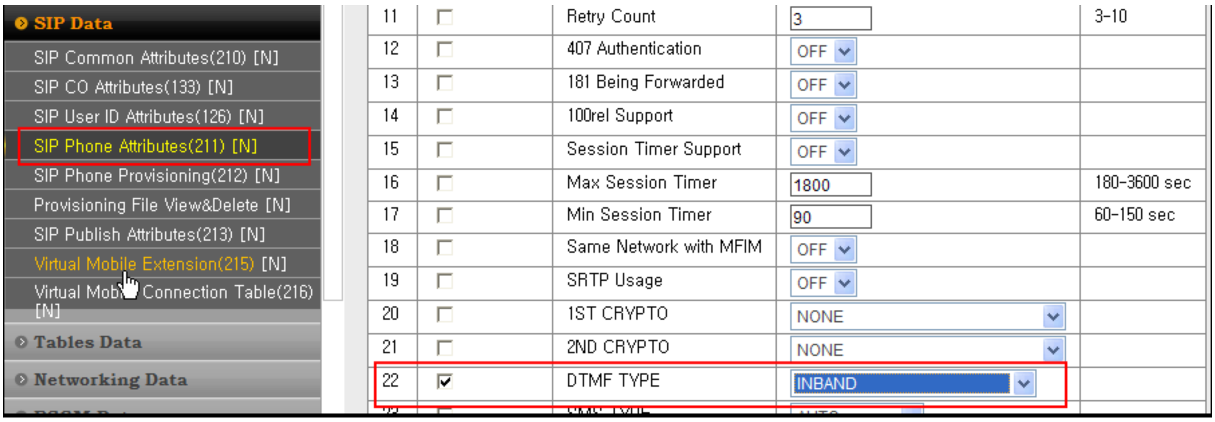
- VMEX RTP is non-recursing by default.





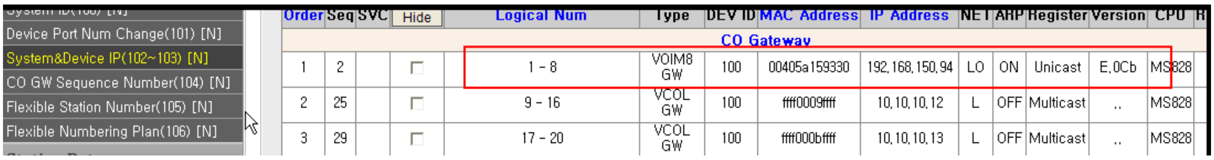
6. DTMF

- Outband signaling by default
- you can set this to INBAND or 2833 in PGM 211 - 'DTMF TYPE'
- If DTMF type is set to INBAND or 2833 then VOIP RTP relay channel is utilized automatically when voice connection is established.



7. DSP implementation for VMEX

- same as normal SIP extension
- VOIP RTP relay channel will be utilized dynamically when need to serve MOH, Ringback/Busy Tone, Paging and so on.
- at least 1/8 to maximum 1/2 of number of concurrent SIP call



Related Features

Hardware

8.18 SIP Virtual CO Line (VCOL)

Description

Virtual channel without DSP usage for SIP trunk.
Voice and Video is direct between end sides.
Support only DTMF type

Operation

1. Incoming call from SIP trunk to system

Voice or Video call from external via SIP Virtual CO line.

2. Make outbound call from system to SIP trunk

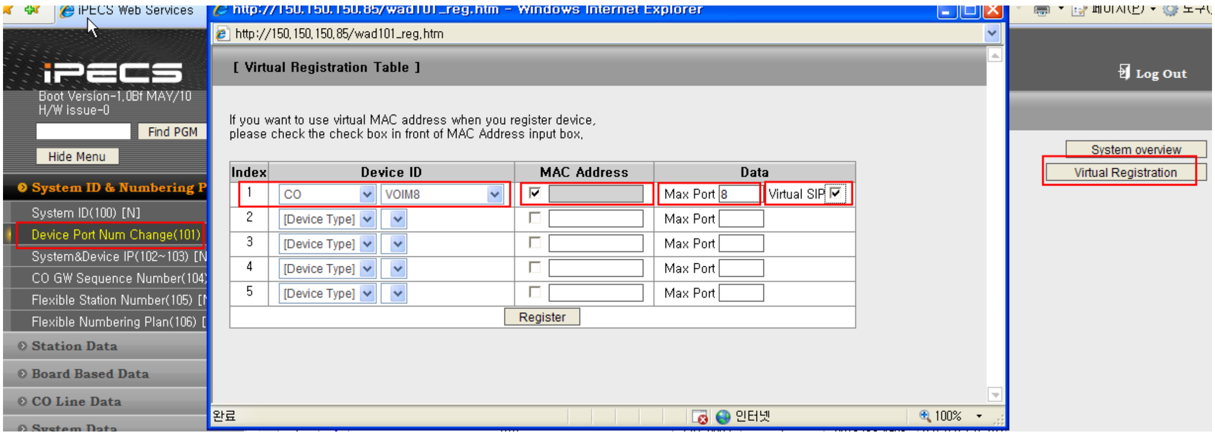
Seize SIP Virtual CO line and make Voice or Video call to SIP trunk.

Conditions

1. Support only Outband DTMF type (SIP Info Message).
2. Tone detection is not supported

Programming

1. Create new SIP Virtual CO lines (VCOL)
   - System ID & Numbering Plans / PGM 101
     'Virtual Registration'
     Device ID : CO / VOIM8 or VOIM24 (you can specify number of channel by 'Max Port')
     MAC Address : check disable and it will be virtually created automatically
     Data : Max port / Virtual SIP ON or not
     - Max Port : number of channels you want to create for virtual SIP CO line channels
     - Virtual SIP : 'should be checked'

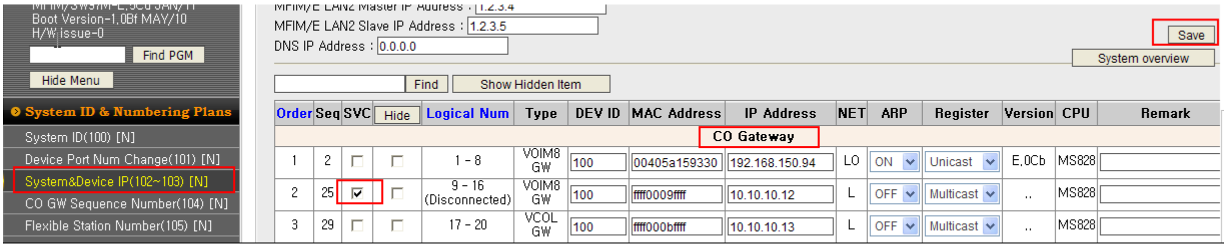


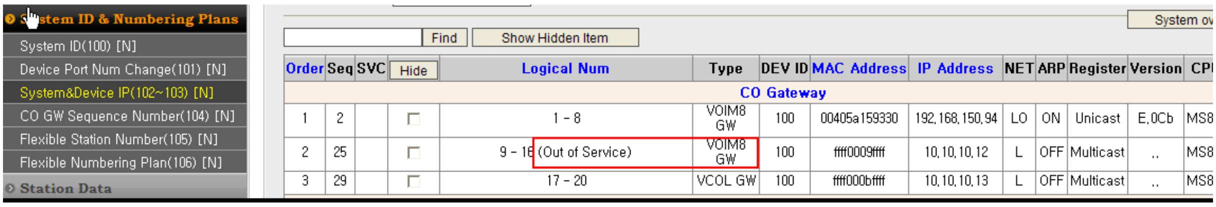
2. Replace a normal SIP CO lines to VCOL
   - the slot should be in 'Out Of Service' state
     (you can force that to 'Out Of Service' state in PGM 103 by click 'CO Gateway', check 'SVC' and save)
   - SIP Data / PGM 133 : set 'Virtual SIP Channel Mode' to 'Yes'

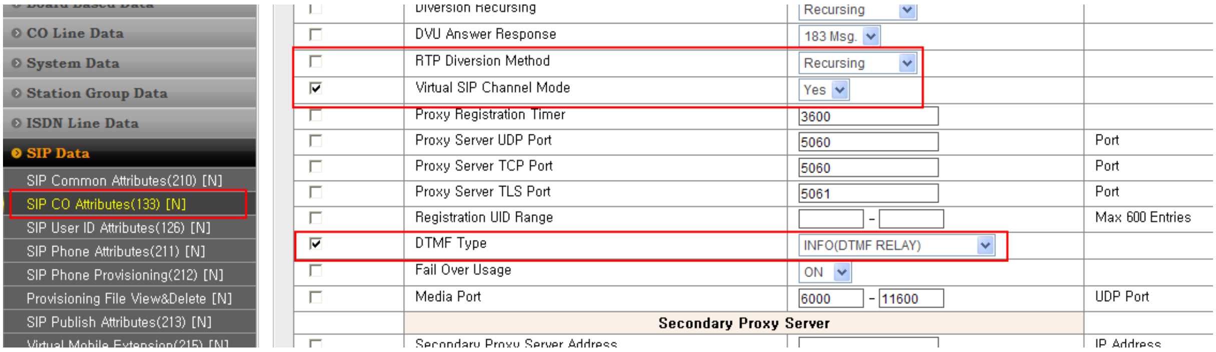
(state is set to activated automatically)
('RTP Diversion Method' is automatically set to 'Non-Recursing')
('CO VoIP Mode' in PGM 141 is set to 'SIP Only' automatically)

c.f) 'RTP Diversion Method' was added in PGM 133. But currently it is only for VCOL.
If we do not need this visible then I will remove or make this display only later

- set 'DTMF Type' in PGM 133 to one of Outband signaling (Outband signaling only, not INBAND and 2833)

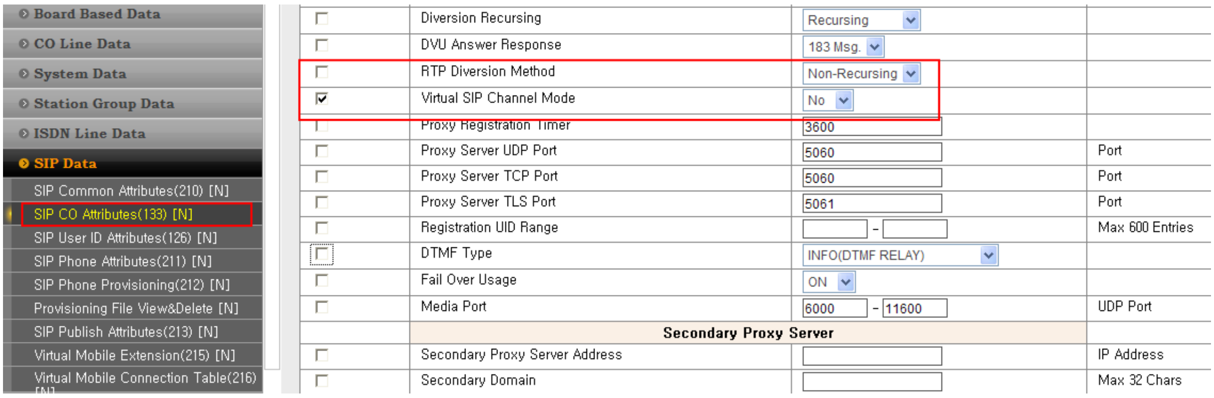






3. Unregister VCOL

- SIP Data / PGM 133 : set 'Virtual SIP Channel Mode' to 'No'
(state is set to disconnected automatically)
('RTP Diversion Method' is automatically set to 'Recursing')
('CO VoIP Mode' in PGM 141 is set to 'COMMON' automatically)
- You can replace type, MAC and IP address for other physical CO slot module.

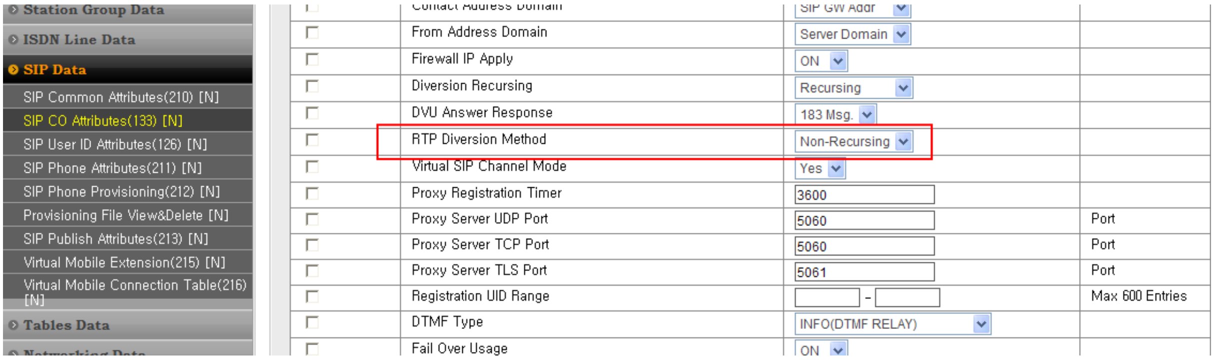


4. VCOL attributes

- Everything is same as a normal SIP CO line channel
- in PGM 133, input Proxy, Domain, and other attributes

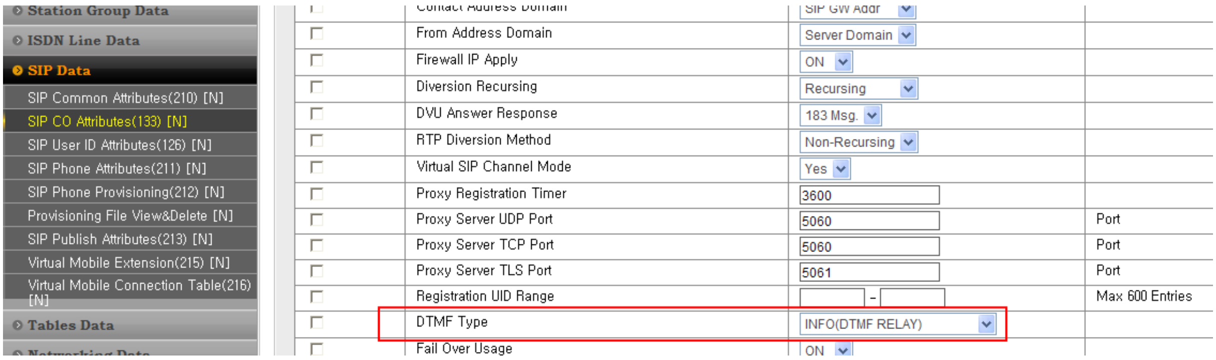
5. RTP Connection

- RTP is Non-Recursing by default and it is forced to be



6. DTMF

- Outband signaling only
- 'DTMF Type' in PGM 133 is 'INFO(DTMF RELAY)' and this is called CISCO type



7. DSP implementation for VCOL

- same as SIP extension
- VOIP RTP relay channel will be utilized dynamically when need to serve MOH, Ringback/Busy Tone, Paging and so on.

- at least 1/8 to maximum 1/2 of number of concurrent SIP call

Related Features

Hardware

iPECS

Feature Operation
&
Description Manual